telecor™ inc.

VS1 Telephone System
Installation, Configuration
& Operating Guide

Intended for use in VS1 Systems
running TVS Software 2.10

June 2003
Man Rev 3

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Manufacturer:
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Patents

Patents Pending

Environmental Specifications

See the Hardware section for specific product specifications.

FCC Information

This equipment complies with Part 68 of the FCC rules. On the bottom of this equipment, there is a label that contains, among other information, the FCC Registration Number and Ringer Equivalence Number (REN) for this equipment. The telephone company may require this information prior to connection of this equipment.

This equipment requires USOC connectors listed below which are provided in the installation of the Telecor VS1 business telephone system.

An FCC compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack which is Part 68 compliant.

The REN is useful to determine how many devices you may connect to a single telephone line and still have all of those devices ring when your number is called. In most, but not all areas, the sum of all RENs per line should be five (5.0) or less. Your local telephone company can verify the maximum REN per line for your calling area.

If this equipment causes harm to the telephone network, the telephone company may notify you in advance that temporary discontinuance of service is required. If advance notice is not practical, the telephone company will notify you of discontinuance as soon as possible. You will also be advised of your right to file a complaint with the FCC.

The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of your telephone equipment. If this happens, the telephone company will provide advance notice so that you can modify your equipment to maintain uninterrupted service.

If the trouble causes harm to the network, the telephone company may ask you to disconnect this equipment from the network until the problem has been solved. If you experience trouble with this telephone equipment, please contact:

Telecor Inc., 1114 Westport Crescent, Mississauga, ON L5T 1G1 Technical Support: (800) 464-3274, Mississauga. www.Telecor.com

You should not try to repair this equipment yourself. Changes or modifications not expressly approved by Telecor Inc. could void your authority to operate the equipment.

This equipment may not be used on public coin service provided by the telephone company. Connection to party lines is subject to state tariffs. (Contact your state public utility commission or corporation commission for information.)

This equipment is hearing aid compatible.

Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at their own expense.

This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

Trade Name(s): Telecor™ VS1™ Business Telephone System
FCC Registration Number: 5NBUSA-32975-MY-E
IC Certification Number: 2403 6616 A
AC REN: (Base System) 1.3B and (T1) N/A
Connectors: (FCC USOC) RJ11C/21X/48 and (IC) CA11A/CA21A/CA81A
Authorized Network Ports: (FCC FIC) 02LS2 04DU9-BN/DN/1KN/1SN/1ZN and (IC) LS/DI1/DI2/DI3

Trade Name: Telecor Cut-Over Box
FCC Registration Number: 5NBUSA-32976-PX-N
AC REN: 1.0B with CID; 0.0B without CID
Connectors: (USOC) RJ-11
Authorized Network Ports: (FIC) 02LS2

Trade Name: Display Phone
FCC Registration Number: FTZCHN-31227-KX-E
IC Certification Number: 2760 7997 A
AC REN: 1.6B
Connectors: (FCC USOC) RJ11C and (IC) CA11A
Authorized Network Port: (FCC FIC) 02LS2 and (IC) LS

Additional Information

This document is current as of the date of publication. Refer to the Telecor Web site, www.telecor.com, for supplemental information.

For TVS Software 2.10
Man Rev: 3
June 2003
E:\VS1 Documents\VS1 ICO\Telecor Manual

CS-03 Information (Canada)

NOTICE: The Industry Canada label identifies certified equipment. The certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent the degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

CAUTION: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority or electrician as appropriate.

NOTICE: The Ringer Equivalence Number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the devices does not exceed 5.

CS-03 Information (Canada)

AVIS: L'étiquette d'Industrie Canada identifie le matériel homologué. Cette étiquette certifie que le matériel est conforme aux normes de protection, d'exploitation et de sécurité des réseaux de télécommunications, comme le prescrivent les documents concernant les exigences techniques relatives au matériel terminal. Le Ministère n'assure toutefois pas que le matériel fonctionnera à la satisfaction de l'utilisateur.

Avant d'installer ce matériel, l'utilisateur doit s'assurer qu'il est permis de le raccorder aux installations de l'entreprise local de télécommunication. Le matériel doit également être installé en suivant une méthode acceptée de raccordement. L'abonné ne doit pas oublier qu’il est possible que la conformité aux conditions énoncées ci-dessus n'empêche pas la dégradation du service dans certaines situations.

Les réparations de matériel homologué doivent être coordonnées par un représentant désigné par le fournisseur. L'enterprise de télécommunications peut demander à l'utilisateur de débrancher un appareil à la suite de réparations ou de modifications effectuées par l'utilisateur ou à cause de mauvais fonctionnement.

Pour sa propre protection, l'utilisateur doit s'assurer que tous les fils de mise à la terre de la source d'énergie électrique, de lignes téléphoniques et des canalisations d'eau métalliques, s'il y en a, sont raccordés ensemble. Cette précaution est particulièrement importante dans les régions rurales.

AVERTISSEMENT: L'utilisateur ne doit pas tenter de faire ces raccordements lui-même; il doit avoir recours à un service d'inspection des installations électriques, ou à un électricien, selon le cas.

AVIS: L'indice d'équivalance de la sonnerie (IES) assigné à chaque dispositif terminal indique le nombre maximal de terminaux qui peuvent être raccordés à une interface. La terminaison d'une interface téléphonique peut consister en une combinaison de quelques dispositifs, à la seule condition que la somme d'indices d'équivalence de la sonnerie de tous les dispositifs n'excède pas 5.

Important Safety Instructions

When using your telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock, and injury to persons, including the following.

1. Read and understand all instructions.
2. Follow all warnings and instructions marked on the product.
3. Unplug this product from the wall outlet before cleaning.
4. Do not use this product near water (for example, in a wet basement).
5. Do not place this product on an unstable cart, stand or table.
6. Slots and openings in the cabinet and the back and bottom are provided for ventilation, to protect it from overheating; these openings must not be blocked or covered. This product should never be placed near a radiator or heat register. This product should not be placed in a built-in installation unless proper ventilation is provided.
7. This product should be operated only from the type of power source indicated in the manual. If you are not sure of the power source to your building, consult your dealer or local power company.
8. Some pieces of equipment have a three-wire grounding type plug, a plug having a third grounding pin. This plug will only fit safely into a grounded power outlet. This is a safety feature. If you are unable to insert the plug into a grounded outlet, contact your electrician. Do not defeat the safety purpose of the grounded plug.

   Some pieces of equipment have a polarized line plug, where one blade of the plug is wider than the other. This plug will fit into the power outlet only one way. This is a safety feature. If you are unable to insert the plug fully into the outlet, try reversing the plug. If the plug still does not fit, contact your electrician. Do not defeat the safety purpose of the polarized plug.
9. Do not allow anything to rest on the power cord. Do not locate this product where the cord will be abused by persons walking on it.
10. Do not use an extension cord with this product's AC power cord. Equipment not associated with the telephone system should not be plugged into the same AC outlet.
11. Never push objects of any kind into this product through the cabinet slots as they may touch dangerous voltage points, or short out parts that could result in risk of fire or electric shock. Never spill liquid of any kind on the product.
12. To reduce the risk of electrical shock, do not disassemble this product. Take it to a qualified service vendor, when service or repair work is required. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock when the product is subsequently used.
13. Unplug this product from the wall outlet and refer servicing to a qualified service vendor under the following conditions:
    A. When the power supply cord or plug is damaged or frayed.
    B. If liquid has been spilled into the product.
    C. If the product has been exposed to rain or water.
    D. If the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions. Improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to normal operation.
    E. If the product has been dropped or the cabinet has been damaged.
    F. If the product exhibits a distinct change in performance.
14. Avoid using a telephone (other than cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
15. Do not use the telephone to report a gas leak in the vicinity of a leak.

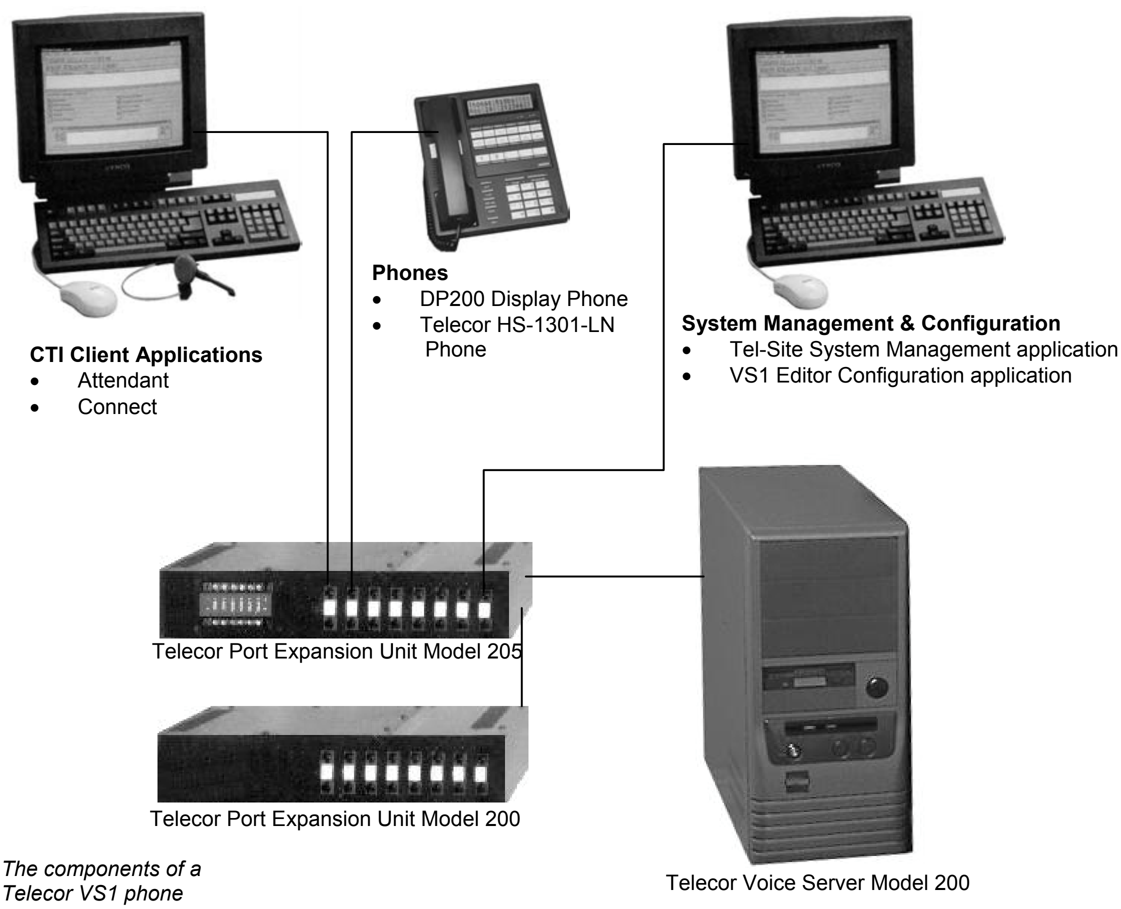
TABLE OF CONTENTS

INTRODUCTION

The Telecor VS1 phone system is a PC-based expandable PBX. The base system has two main components: the Telecor Voice Server (TVS) and the Telecor Port Expansion Unit (PEUs). Each PEU provides 16 ports. Up to 11 additional PEUs can be added with 16 ports each, enabling the Telecor VS1 phone system to expand to 192 ports, depending upon the type of station sets used.

A number of station options are available in the form of Computer Telephony Integration (CTI) applications or desk phones. The system is managed with the Tel-Site system management application, and configured with the VS1 Editor configuration application.

The diagram below shows the main components of the Telecor VS1 phone system.



The components of a Telecor VS1 phone system

Using this Guide

This guide is designed to instruct Telecor Value Added Resellers how to install and configure the VS1 phone system. Follow the instructions and recommendations provided to ensure proper installation and operation of the Telecor VS1 phone system. This guide is organized into six sections to address specific areas of the Telecor VS1 phone system.

Section 1 introduces the VS1 hardware and provides installation and configuration instructions for the hardware.

Section 2 provides an overview of the Tel-Site system management application, and details the different connection methods available (on-site and off-site) for connecting to a customer site.

Section 3 gives instructions on using the VS1 Editor configuration application to configure a VS1 phone system.

Section 4 introduces the different station options available, along with quick operating instructions and voice mail features.

Section 5 provides extensive Reference information, such as Station Message Detail Recording (SMDR) information. Also includes a glossary of terms.

Section 6 provides a comprehensive Index.

Conventions in this Guide

This guide is designed to help you install and configure the *Telecor* VS1 phone system quickly and efficiently. There are some words and symbol conventions used throughout the guide to help you along the way.

- If you are not at the system site (off-site) or are on a remote computer, and are using the Tel-Site system management and VS1 Editor configuration application, you are asked to **Click** certain buttons with the mouse.

- If you are configuring the system at the system site (on-site) using a monitor and keyboard, and the guide directs you to select a command or selection, you need to press the UP/DOWN/RIGHT/LEFT arrows, or press TAB, on the keyboard to move to that selection or command, and then press ENTER to execute the command or make the selection.

- If you are asked to **Type**, it means you are to use the keyboard on your computer.

- You are asked to **Press** certain buttons on the phone, or keyboard, to perform a series of steps.

- When you see a graphic in the left margin, it means you are to click the corresponding button on the window.

Font Styles Used

Bold	Elements found on the Tel-Site, VS1 Editor, Telecor Attendant, and Telecor Connect application screens, such as windows, menus, dialog boxes and commands.
SMALL CAPS	Keys on the keyboard or phone.
Italics	Image captions.
Note	Important or additional information.
`command`	Command-line syntax or file names.
`argument`	Indicates an argument for which you must supply a value.
`{x}`	An argument or a constant within braces { } is required. Do not type the braces when entering the value.
`[x]`	An argument or a constant within square brackets [ ] is optional. Do not type the brackets when entering the value.
`x|y|z`	Constants or arguments separated by a vertical bars requires a choice.
RESET REQUIRED!	The Telecor Voice Server (TVS) must be reset in order for configuration changes to take effect.
WARNING	**Warning!** Instructions that must be followed to prevent damage to the system.

Hardware

HARDWARE OVERVIEW

This section of the ICO Guide describes the following hardware components of the VS1 Telephone System.

- Telecor Voice Server Model 200 (PV-CSU-200)
- Port Expansion Unit Model 250 (PV-PEU-250)
- Port Expansion Unit Model 205 (PV-PEU-205)
- Port Expansion Unit Model 200 (PV-PEU-200)
- Dry Contact Unit Model 100 (PV-DCU-100)
- Cut-Over Box (PV-HWC-P10)
- 32-Port Host Adapter Card (PV-HA2-032)
- 64-Port Host Adapter Card (PV-HA2-064)
- Caller ID Option Module (PV-HWC-C10)
- 10Base-T Network Interface Card (PV-HWC-E10)
- T1 Interface Card (PV-HWC-T10)
- TVS VGA Video Card (041-000-0013)
- SC200 Switch Card (701-000-0044)
- Computer Telephony Interface Module (CTIM) (PV-HCT-C01)
- PC Option Module (PCOM) (PV-HCT-M01)
- ACD Status Board Option Kit (PV-HWC-A10)

Telecor Voice Server (TVS) Model 200

The Telecor Voice Server Model 200 (PV-CSU-200) is the central component of the Telecor VS1 telephone system. It houses the proprietary software and hardware that operates the system. The TVS is a commercial, industrial-grade server that uses industry-standard hard drives and can store at least 30 hours of voice messages. It comes equipped with a 56 Kbps modem for Remote System Access (RSA), enabling you to make fast, reliable programming changes. The modem card is identified by its two RJ-11 adapters. Only one modem can be installed in the TVS.

The TVS comes with a 32-port Host Adapter Card already installed. You can install a maximum of four Host Adapter Cards in the TVS. *For information on installing additional Host Adapter Cards, see "Installing a Host Adapter Card," page 35.*

Telecor Voice Server

Hardware Specifications

Dimensions (WxHxD): 5.75" x 14.75" x 13.75"

Weight: 27 lbs.

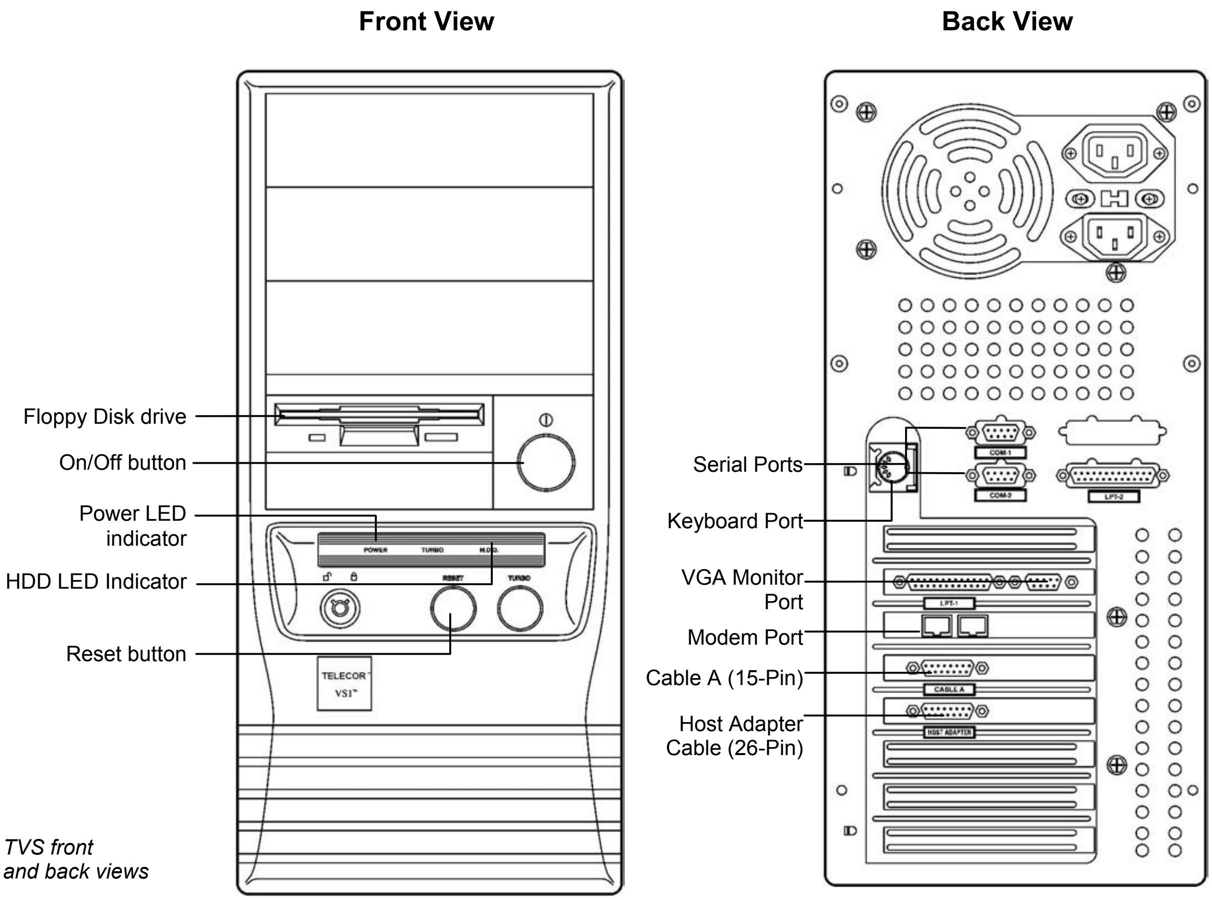
Power Requirements: 120 VAC

Temperature: 4°–38° C operational;

Humidity: 5%–95% non-condensing

Standards Compliance: FCC Part 15; UL1459

Power Consumption: max. 240VA



TVS front and back views

I/O Port and IRQ Usage

The following tables list the I/O Port and Interrupt Request settings used by the Telecor VS1 phone system.

IRQ	Usage
0	System Timer
1	Keyboard
2	Cascade to second programmable Interrupt Controller
3	COM 2
4	COM 1
5	COM 3 (Internal RSA modem)
6	Floppy Drive
7	LPT 2
8	Real-time Clock/Calendar
9	SC200 Switch Card
10	Available
11	Available (Default setting for Network Interface Card)
12	Reserved for PS/2 mouse port on current system board
13	Numeric Coprocessor
14	IDE Controller
15	SC200 Switch Card

Port Address*	Usage
210–21F	System Timer
2F8–2FF	COM 2
300–307	HA200 Host Adapter (ports 1–32)
308-30F	HA200 Host Adapter (ports 33–64)
310–317	HA200 Host Adapter (ports 65–96) or T1 Interface Card 1
318–31F	HA200 Host Adapter (ports 97–128) or T1 Interface Card 0
320–327	HA200 Host Adapter (ports 129–160)
328–32F	HA200 Host Adapter (ports 161–192)
340–35F	Network Interface Card
379–37B	LPT 1
3E8–3EF	COM 3
3F8–3FF	COM 1

*All port addresses are listed in hexadecimal format.

Port Expansion Unit (PEU)

Each VS1 business telephone system requires one or more Port Expansion Units (PEUs) to which all standard CO lines and station equipment are connected. Each PEU has 16 RJ-11 ports.

The first PEU is attached to the Telecor Voice Server (TVS). An additional PEU can be daisy-chained off the first PEU. The TVS comes with a 32-port Host Adapter Card from which a maximum of two PEUs can be daisy-chained. To connect additional PEUs to the TVS, you must install additional Host Adapter Cards. *For more information on installing a Host Adapter Card, see "Installing a Host Adapter Card," on page 35.* A maximum of 12 PEUs can be connected to the TVS for a total of 192 ports.

Three PEU models are available as described below:

PEU 250

The Port Expansion Unit Model 250 (PV-PEU-250) contains 16 generic ports that you can configure as CO or station ports. Ports are configured using pushbutton switches located next to each port jack. Eighteen dual-color LEDs (amber and green) on the front of the PV-PEU-250 indicate power and communication status. The PV-PEU-250 includes integrated caller ID circuitry on all 16 ports. Automatic cut-over is provided for ports 1 through 4 if the ports are configured as CO ports. If a power failure occurs, the ports cut over to ports 5 through 8 respectively, if ports 5 through 8 are configured as extensions. The PV-PEU-250 can be used as any PEU in a system. It does not contain external device contacts. External devices are connected to TVS via the Dry Contact Unit Model 100 (PV-DCU-100).

Note	If your current TVS software version is prior to 2.9, only the first eight ports on the Model 250 are configurable as CO lines.

PEU 205

The Port Expansion Unit Model 205 (PV-PEU-205) contains six external device contacts including two zone pager outputs for overhead paging systems, two music inputs for music on-hold or promotions on-hold, and two dry contact relays to attach electronic door locks, buzzers, or sirens. The Model 205 typically serves as the first PEU on a system with Model 200 PEUs connected as the second and additional PEUs. Of the 16 ports the first eight ports can be configured as CO ports or station ports. The remaining eight ports can be configured as station ports only.

PEU 200

The Port Expansion Unit Model 200 (PV-PEU-200) has all the same features of the Model 205 but it does not have the six external device contacts. The Model 200 typically serves as the second, and additional PEU on systems using the Model 205 PEU as the first PEU. Of the 16 ports, the first eight ports can be configured as CO ports or station ports. The remaining eight ports can be configured as station ports only.

PEU 250

PEU 205

PEU 200

Hardware Specifications

Dimensions (WxHxD): 16.5" x 3" x 13.5"

Weight: 27 lbs.

Power Requirements: 120 VAC

Temperature: 4°–38° C operational;

Humidity: 5%–95% non-condensing

Standards Compliance: FCC Part 15, FCC Part 68; UL1950

Power Consumption: max 120VA

Dry Contact Unit (PV-DCU-100)

The Dry Contact Unit Model 100 (PV-DCU-100) is used to connect external contacts in systems that use Model 250 Port Expansion Units. The PV-DCU-100 provides connections for two speakers (zones), two music inputs, and two dry contact (relay) outputs. It is connected to the switch card in the TVS.

PV-DCU-100

Hardware Specifications

Dimensions (WxHxD): 16.5" x 1.75” x 2.75”

Weight: 1 lb. 6 oz.

Power Requirements: No power required

Temperature: 4°–38° C operational;

Humidity: 5%–95% non-condensing

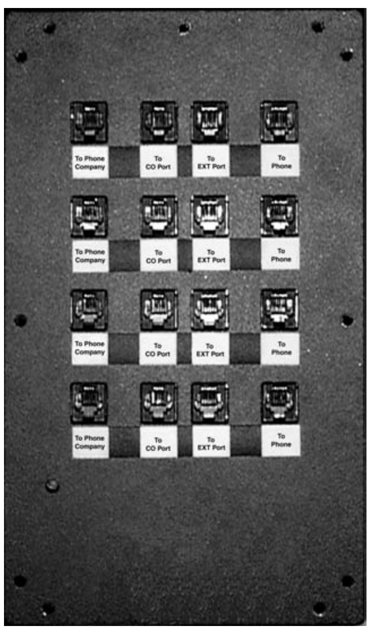
Standards Compliance: FCC Part 15, FCC Part 68; UL1950

Cut-Over Box

An optional Cut-Over Box provides power failure transfer for up to four analog CO lines by routing the CO lines directly to station sets on the VS1 telephone system. If you have a power failure or if the Cut-Over Box is turned off, each station connected to the Cut-Over Box becomes a direct outside line for incoming and outgoing calls. All phones used with the VS1 phone system can be connected to a Cut-Over Box. Although the Cut-Over Box is an optional component, it is recommended for all VS1 phone systems.

Note	If your VS1 system uses the Port Expansion Unit Model 250, you do not need a separate cut-over box. Cut-over functionality is built into the PEU Model 250.

Note	When installing a new system, Telecor recommends that the Cut-Over Box cable connections are made after the system has been installed, the software configured, and all stations have been tested and verified. This aids in determining which stations will be connected to the Cut-Over Box.



Cut-Over Box

Hardware Specifications

Dimensions (WxHxD): 5.75" x 9.75" x 3.7"

Weight: 5.5 lbs.

Power Requirement: 12 Vdc provided by included 120 VAC adapter

Temperature: 4°–38° C operational

Humidity: 5%–95% non-condensing

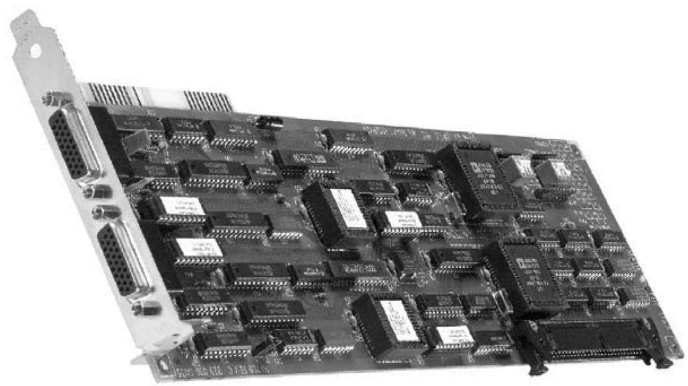
Adapter Specifications: RJ-11

Standards Compliance: FCC Part 15; UL1459

Host Adapter Cards

A Host Adapter Card is used to connect PEUs to the TVS. Each TVS comes with one 32-port Host Adapter Card installed, meaning that it can support two 16-port PEUs. Additional Host Adapter Cards can be installed in the TVS to increase the port capacity of the phone system.

Aside from the 32-port model Host Adapter Card, Telecor offers a 64-port model. The 64-port Host Adapter Card supports four PEUs, giving you 16 ports per PEU for a total of 64 ports. Installing three 64-port Host Adapter Cards expands the number of ports to 192.



64-Port Host Adapter Card

Hardware Specifications

Dimensions: 13.5" x 4"

Weight:
32-Port Host Adapter Card: 9 oz.
64-Port Host Adapter Card: 10.3 oz.

Power Requirement: + 5 Vdc from ISA Bus

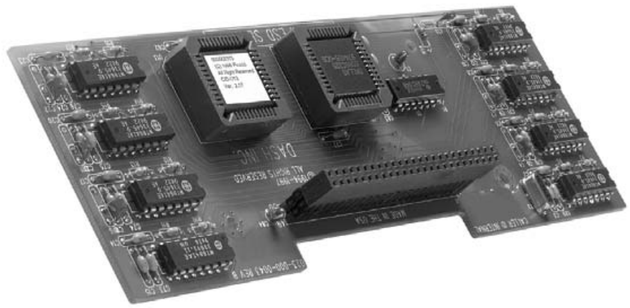
Adapter Specifications:
32-Port Host Adapter Card: One DB-26 Connector
64-Port Host Adapter Card: Two DB-26 Connectors

Architecture: 8-Bit Industry Standard Architecture (ISA)

Standards Compliance: FCC Part 15; UL1459

Caller ID Option Module

The Caller ID Option Module permits Caller ID (provided by local phone company) to work on the Telecor VS1 phone system. The Caller ID Option Module requires Voice Server Software Version 2.7 or later. The Caller ID Option Module is installed inside a Port Expansion Unit Model 200 (PV-PEU-200) or Port Expansion Unit Model 205 (PV-PEU-205). The Caller ID module is not used with the Port Expansion Unit Model 250 (PV-PEU-250). The Model 250 contains integrated Caller ID circuitry on all 16 ports.



The Caller ID Option Module

Hardware Specifications

Dimensions: 7.25" x 2.5"

Weight: 3 oz.

Power Requirements: +5 Vdc and –5 Vdc, supplied by the PEU

Standards Compliance: FCC Part 68, UL1950

Adapter Specifications: 50-pin header connector

Architecture: Two-layer PCB

10Base-T Network Interface Card

Installing a 10Base-T Network Interface Card enables Station Message Detail Recording (SMDR) data to be written a network server. The 10Base-T Network Interface Card must be installed in the Telecor Voice Server (TVS) if you want to log on to a Novell® network or an NT 4.0 server running File and Print Services for Netware (FPNW).

10Base-T Network Interface Card

Hardware Specifications

Dimensions: 6.25" x 2.75"

Weight: 3.5 oz.

Power Requirement: + 5 Vdc from ISA Bus

Adapter Specifications: RJ-45 connector

Architecture: 16- Bit Industry Standard Architecture (ISA) bus

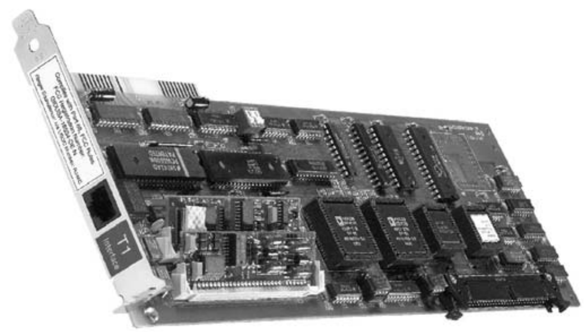
Standards Compliance: FCC Part 15 Class A

Compatibility: Novell® NE2000

T1 Interface Card

T1 is a digital transmission link with the capacity of 1.544 Megabits per second (Mbps). T1 uses two pairs of normal twisted wires, and can normally handle 24 simultaneous voice conversations. The T1 Interface Card is installed inside the TVS and supports 24 T1 channels. The T1 Interface Card supports E&M, DID, DNIS, ANI, Ground Start, and Loop Start. A maximum of two T1 Interface Cards can be installed in the TVS. The following table shows the number of T1 channels and Analog CO lines that can be configured for each T1 card installed in the TVS.

Number of T1 Cards	T1 channels	Maximum # of PEUs	Maximum Analog CO lines	Maximum Total Ports
0	0	12	96	192
1	24	10	72	184
2	48	8	48	176



T1 Interface Card

Hardware Specifications

Dimensions: 13.5" x 4"

Weight: 10 oz.

Power Requirement: +5 Vdc from ISA Bus

Adapter Specifications: RJ-48

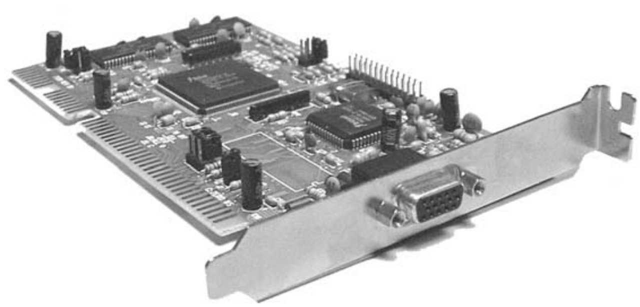
Architecture: 8-bit Industry Standard Architecture (ISA)

Standards Compliance: FCC Part 15; UL1459

TVS VGA Video Card

The Telecor Voice Server (TVS) Model 200 includes a built-in VGA video card.

Note	Even if you do not plan to connect a monitor to the system, the video card is required for operation and must remain installed.



VGA Video Card

Hardware Specifications

Dimensions: 6.25" x 3.75"

Weight: 3.2 oz.

Power Requirement: +5 Vdc from ISA Bus

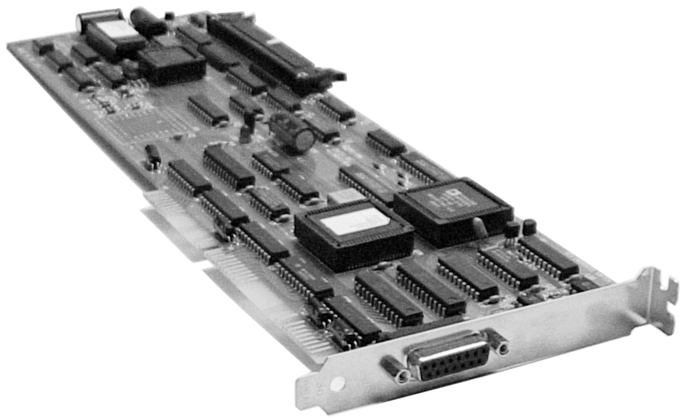
Adapter Specifications: DB15 connector

Architecture: 16-Bit Industry Standard Architecture (ISA)

Standards Compliance: FCC Part 15

SC200 Switch Card

Each TVS comes with a preinstalled Switch Card, identified by its 15-pin connector labeled Cable A. The primary functions of the Switch Card are to support call switching, zone paging, external music on-hold sources, relay contacts, and the voice channels. It has a connection for an external fan with a fan fault monitor. In addition it provides for monitoring of the case temperature. The switch card is also responsible for processing system resets.



SC200 Switch Card

Hardware Specifications

Dimensions: 13.5" x 4"

Weight: 9 oz (258 grams)

Power Requirement: + 5 Vdc and (+12 Vdc from ISA Bus if external fan is connected)

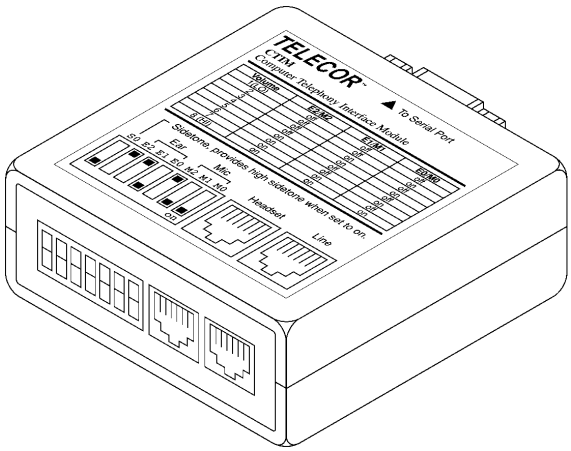
Adapter Specifications: DB15 connector

Architecture: 16-Bit Industry Standard Architecture (ISA)

Standards Compliance: FCC Part 15; UL1459

Computer Telephony Interface Module (CTIM)

The Computer Telephony Interface Module (CTIM) provides a digital interface to the VS1 phone system for the Telecor Attendant and Connect applications. The CTIM is an external device with a headset jack intended for use only with the VS1 phone system.



CTIM

Hardware Specifications

Dimensions: 2.6" x 2.6" x 1.1"

Weight: 2.8 oz.

Power Requirements: Powered by PEU

Temperature: 10°–50° C operational

Humidity: 5%–95% non-condensing operational

Adapter Specifications: DB-9 Serial Port; RJ-11 Line; RJ-22 Headset

PC Option Module (PCOM)

The PC Option Module (PCOM) provides an externally connected serial interface to a computer for data sent by the Telecor Voice Server (TVS). The PCOM is used with the Telecor Connect CTI client application. By installing a PCOM, Telecor Connect users can use their computer screen for call processing and keep the phone on their desk. Keeping a phone on the desk also allows calls to be received when the computer is turned off.

PC Option Module (PCOM) (cable not pictured)

Hardware Specifications

Dimensions: 3.25" x 2.5"

Weight: 3 oz.

Power Requirements: Powered by PEU

Temperature: 10°–50° C operational

Humidity: 5%–95% non-condensing operational

Adapter Specifications: RJ-11 connectors and one DB25 connector

ACD Status Board

The ACD Status Board provides a display of selected ACD groups and their status. The ACD Status Board can be hung on a wall or placed on a solid surface at eye-level for easy viewing. The ACD Status Board allows supervisors to monitor the progress of calls through the ACD and make necessary changes quickly to ACDs requiring additional agents. The ACD Status Board can be customized to display a single ACD group, scroll through all active ACD groups, or display no ACD groups.

Installation and setup of the ACD Status Board is quick and straightforward. No additional hardware or software needs to be installed on the Telecor Voice Server (TVS) or the Port Expansion Unit (PEU). The ACD Status Board is connected to the VS1 business telephone system through the ACD Status Phone. The ACD Status Phone is functionally identical to the Display Phone Model 200 (DP200), except that the display is customized to monitor ACD status. The ACD Status Board serves as an external monitoring display of the ACD Status Phone.

The ACD Status Board Option Kit includes the ACD Status Phone and an ACD Status Board connection cable. The ACD Status Board itself must be purchased from Spectrum, the Board's manufacturer, or from a Spectrum dealer.

ACD Status Board and ACD Status Phone

ACD Status Board Manufacturer

Product Number: **DAS-1512R1**
Spectrum
10048 Easthaven Blvd.
Houston, TX 77075
800.392.5050
Please contact Spectrum or a Spectrum authorized dealer, to purchase the Board. This product is not available from Telecor Inc.

HARDWARE CONFIGURATION & INSTALLATION

This section contains instructions and figures to configure and install the VS1 phone system hardware.

The Telecor VS1 telephone system can be either rack mounted or wall mounted. Each PEU comes with brackets for a wall mount and rack mount setup. Mounting hardware for other components is ordered depending if a wall mount or rack mount setup is used. Prior to mounting the equipment, the PEUs must first be considered to support CO Lines or Station Extensions.

Configuring the PEU 250

The Model 250 PEU has 16 ports, each of which can be configured as a CO or Station port. It is recommended that the PEU 250 serve as the first PEU on the system, as it is labeled with ports 1 to 16. Ports are addressed sequentially by the TVS. When configuring more than one PEU, verify that each port is correctly identified before changing switch settings.

To configure the PEU 250:

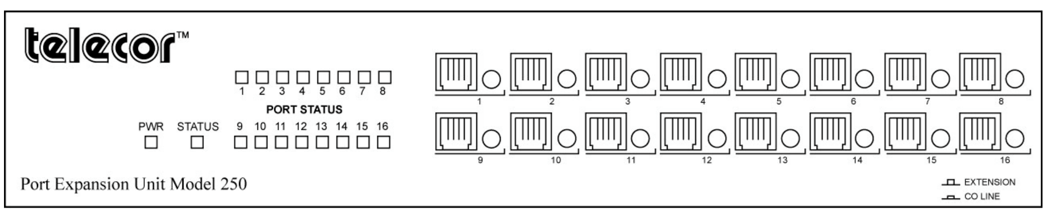
1. Use the pushbutton switches next to each port jack to set the port as a CO Line or Extension.

   - CO Line = button in

   - Extension = button out

Note: If your current software version is prior to 2.9, only the first eight ports on the PEU Model 250 are configurable as CO lines.

Note: In order for Caller ID to work on Ports 1 - 8, Port 1 must be set as a CO Port. In order for Caller ID to work on Ports 9 – 16, Port 9 must be set as a CO Port.



Front view of PEU 250

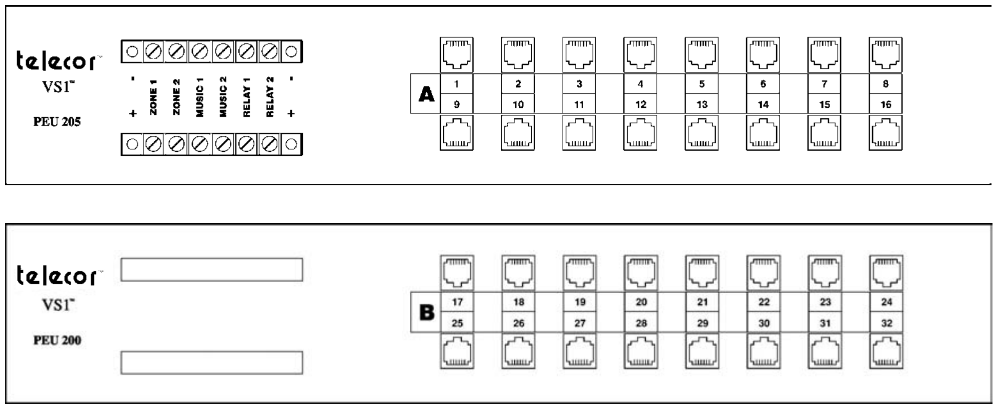
Cut-Over Functioning on the PEU 250

The PEU 250 provides power failure cut-over functioning for up to four analog CO lines by routing the CO lines directly to station sets on the VS1 telephone system. A Port from 1 through 4 configured as a CO Line will cut over to an associated Port from 5 through 8, respectively, if configured as an Extension.

Port configured as CO Line	Cuts over to Port if configured as Extension
1	5
2	6
3	7
4	8

Configuring the PEU 205 and 200

Each PEU 205 and PEU 200 has 16 ports. Ports 1–8 can be configured as either CO lines or stations. Ports 9–16 are for stations only. Ports are addressed sequentially by the TVS. When configuring more than one PEU, verify that each port is correctly identified before changing switch settings. It is recommended that the PEU 205 serve as the first PEU on the system.



PEU 205 Ports 1-16 and PEU 200 Ports 17-32 (front view)

To configure the PEU 205 or PEU 200:

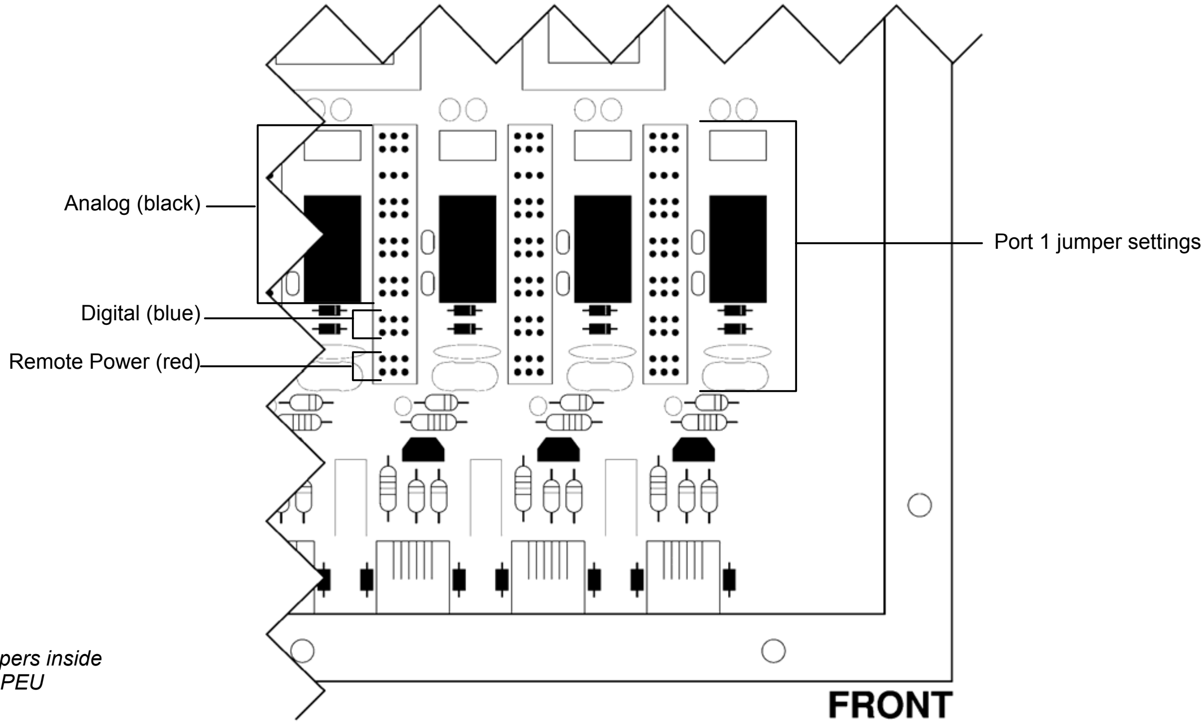
1. Attach the correct PEU labels between the RJ-11 jacks on the front of each PEU. Label the first PEU with ports 1-16 and the second PEU with ports 17–32, and so on. It is recommended that the PEU 205 serve as the first PEU on the system.

2. Remove the top panel on the PEU.

WARNING! Do not remove the smaller (top) panel, which covers the power supply. The power supply area contains hazardous voltages and has no adjustable settings.

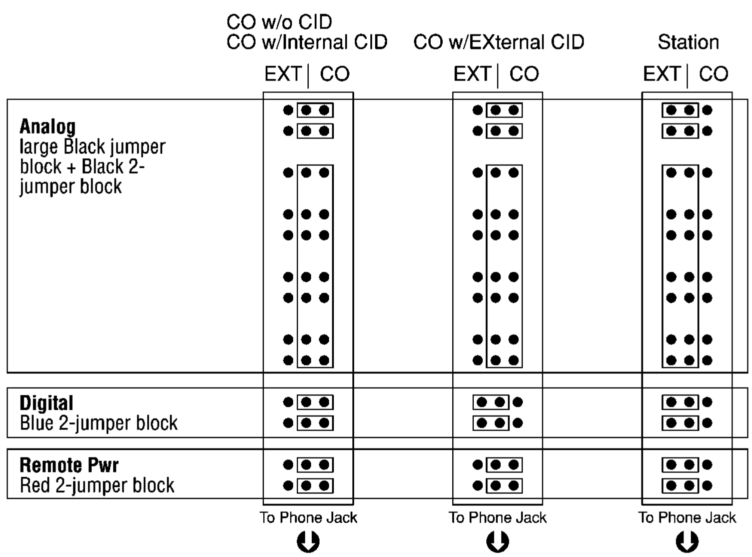
3. Locate the station/CO jumper sets for Port 1. There are eight blocks of pins. Each contains black blocks for analog, blue blocks for digital, and red blocks for remote power.

   - Ports 1–8 have both station and CO jumper sets.
   - Ports 1–4 are preset as CO ports and Ports 5–8 are set as station ports. Up to four additional CO ports can be configured by repositioning the jumpers on Ports 5–8.
   - Ports 9–16 do not have jumpers.



Jumpers inside the PEU

4. Set the jumpers for Port 1 based on the diagram below. Move all jumpers to the left side for a station. Move all Analog (black) and Remote Power (red) jumpers to the right side for CO ports, and set the digital (blue) jumpers based on having Internal Caller ID or External Caller ID.



Jumper Settings

5. Repeat Steps 3 and 4 to set the jumpers for Ports 2–8.

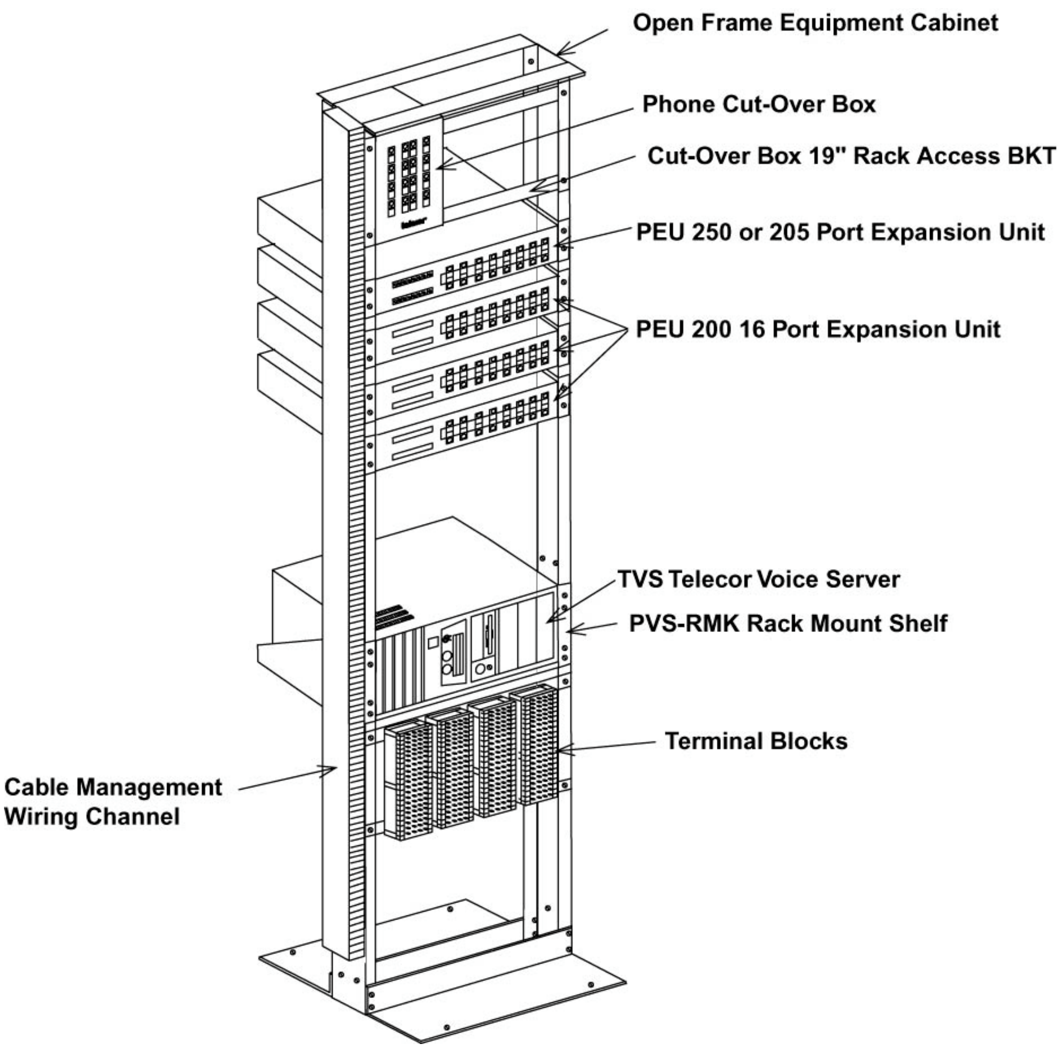
6. When finished, replace the panel on the PEU.

7. Repeat Steps 1-6 for additional PEUs.

Rack Mount Setup

The Phone Cut-Over Boxes are located at the top of the rack. The PEU-250 or 205 is located directly below the Cut-Over Box, followed by the PEU-200 units. The CO lines from the Telco Demarc terminate onto the Cut-Over Boxes. Connections are then made with RJ11 patchcords from the Cut-Over Box to the CO Ports on the PEU-250 or PEU 205 *(see “Connecting the Cut-Over Box,” page 33)*. If your installation does not include Cut-Over Boxes, then simply connect the CO lines to the designated CO ports on the PEU 250 or PEU 205.

Station field wiring and Telco Demarc lines are terminated onto TM-16X3 or TM-8X3 terminal blocks *(see “Terminal Blocks,” page 25).* The terminal blocks may be located on the equipment rack.



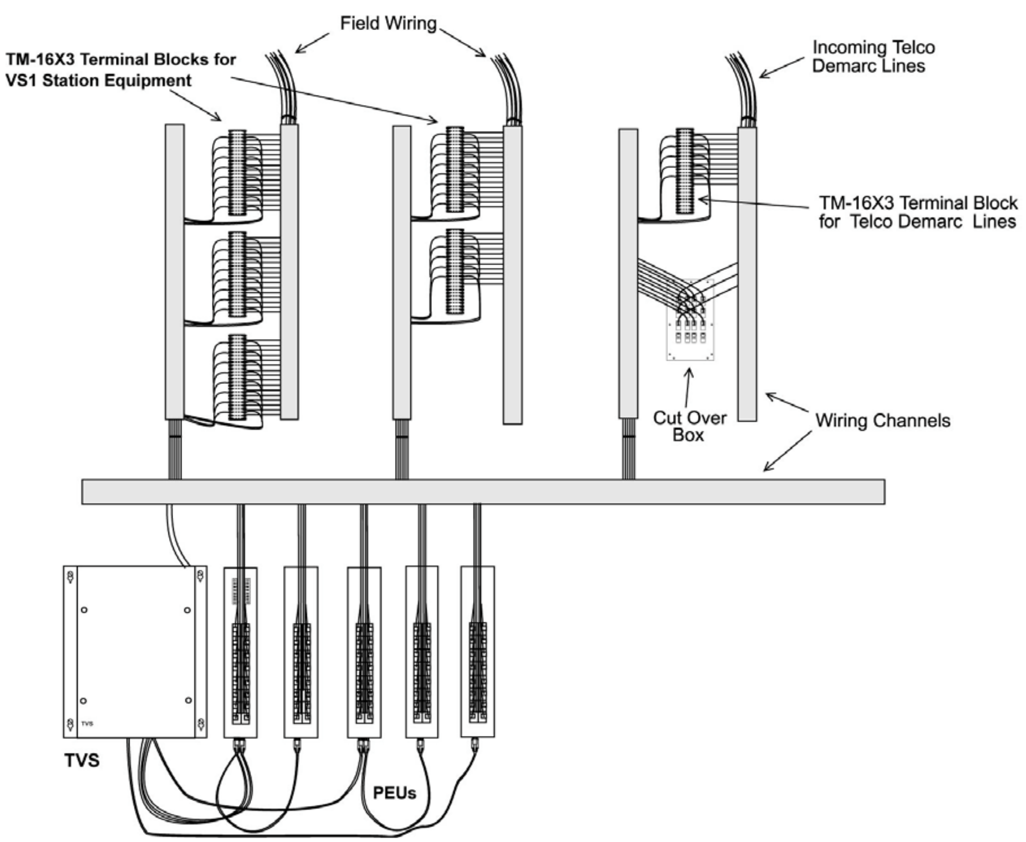
The TVS is located at the bottom portion of the rack and conveniently mounts into a PVS-RMK Rack Mount Shelf. The TVS must be located no more than three feet away from the PEU-250 or PEU 200.

Wall Mount Setup

The Phone Cut-Over Boxes are located adjacent to the incoming Telco Demarc lines, which terminate onto the Cut-Over Boxes. Connections are then made with modular patchcords from the Cut-Over Box to the CO Ports on the PEU-250 or PEU 205 *(see "Connecting the Cut Over Box," page 33)*. If your installation does not include Cut-Over Boxes, then simply connect the CO Lines to the designated CO ports on the PEU-250 or PEU 205.

Station field wiring and Telco Demarc lines are terminated onto TM-16X3 or TM-8X3 terminal blocks *(see "Terminal Blocks," page 25)*. Telecor recommends that the terminal blocks be arranged in an orderly manner on a plywood sheet mounted onto the wall. **DO NOT MOUNT ANY EQUIPMENT DIRECTLY ON DRYWALL SURFACES.** Wiring channels should be used to neatly house all the field wiring.

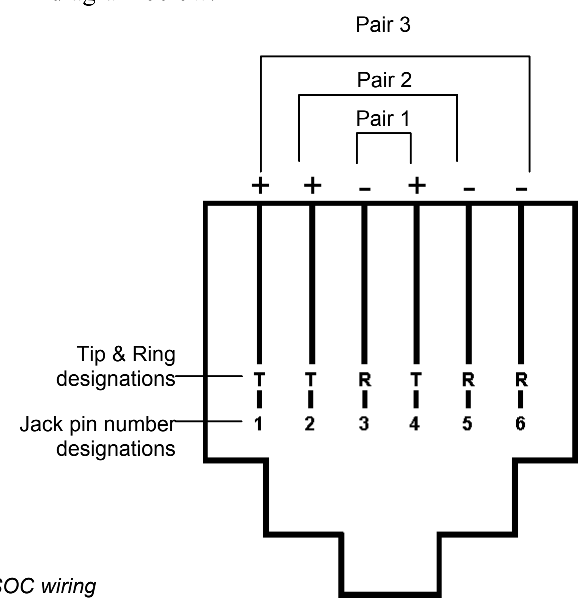
The TVS must be located no more than three feet away from the PEU-250 or PEU-205.



PEU Wiring Requirements

Each port on the PEU supplies remote power to display phones, and requires the use of 3-pair wiring to connect the station to the PEU. For all other extensions, the PEU requires 2-pair wiring. All ports are universal, enabling you to use any station option.

USOC is the wiring standard that should be followed for installing the VS1 phone system. Wiring should adhere to the color code for USOC 6 conductor telecommunication lines as shown in the diagram below.



Cable	Jack
White w/ blue stripe	Green
Blue w/ white stripe	Red
White w/ orange stripe	Black
Orange w/ white stripe	Yellow
White w/ green stripe	White
Green w/ white stripe	Blue

Pair	Signal
Pair 1	Analog/Voice
Pair 2	Digital
Pair 3	Remote Power

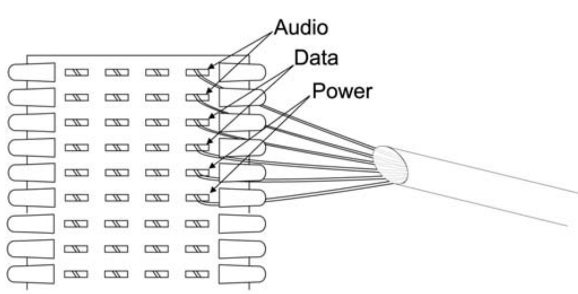
USOC wiring

Terminal Blocks

The Prewired 66 Blocks with Modular Telephone Jacks should be used for interconnection between the RJ11 jacks of the PEUs and the station field wiring and Telco Demarc lines. Each CO or station line from the terminal block is connected to the corresponding CO or station port on the PEU with an RJ11 patchcord. Two versions of termination blocks are available:

- TM-8X3 contains 8 RJ11 telephone jacks for connection of 8 stations or CO lines. Only one side of the termination block is used. The other side is unconnected and can be used for cross connects.

- TM-16X3 contains 16 RJ11 telephone jacks for connection of 16 stations or CO lines. Both sides of the termination block are used. This conveniently corresponds with a 16-port PEU.



Terminal Block Wiring

Connecting the Telecor Voice Server (TVS)

TVS to PEU 250

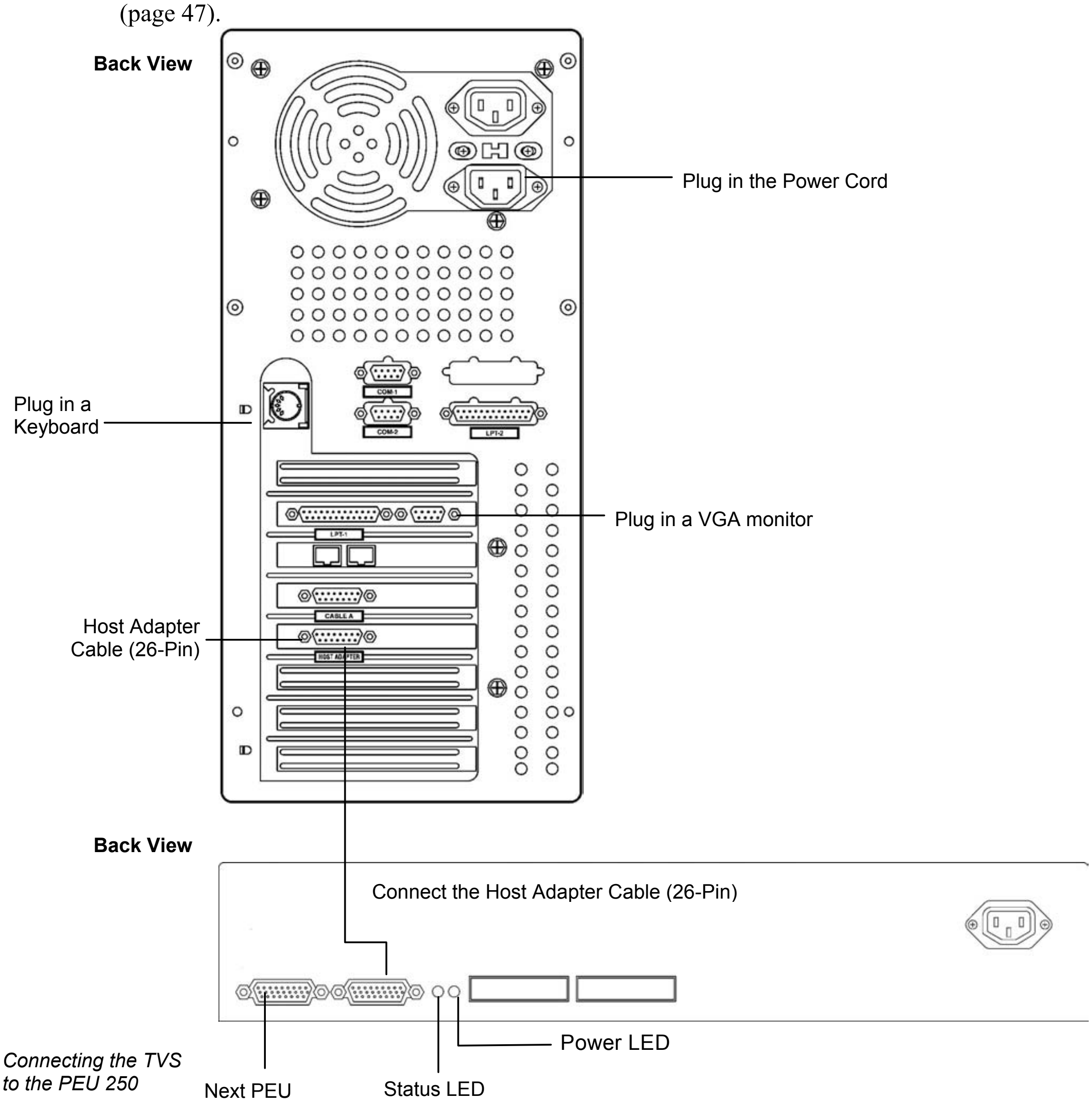
The diagram shows the correct port for each cable on the TVS and PEU 250. To connect the TVS to the first PEU 250, complete the following steps.

1. Plug one end of the 26-pin data cable in the Host Adapter port on the TVS; plug the other end of the 26-pin data cable in the Host Adapter port on the PEU.

2. Plug the Power cord into the TVS, and then connect it to a power strip.

   - The Power LED on the PEU 250 is lit green to indicate that the power is on.
   - The Status LED on the PEU 250 is lit green to indicate that the PEU is connected to the TVS.

3. Plug in a keyboard and VGA monitor. This is required to create an emergency boot floppy (page 47).



Connecting the TVS to the PEU 250

TVS to PEU 205

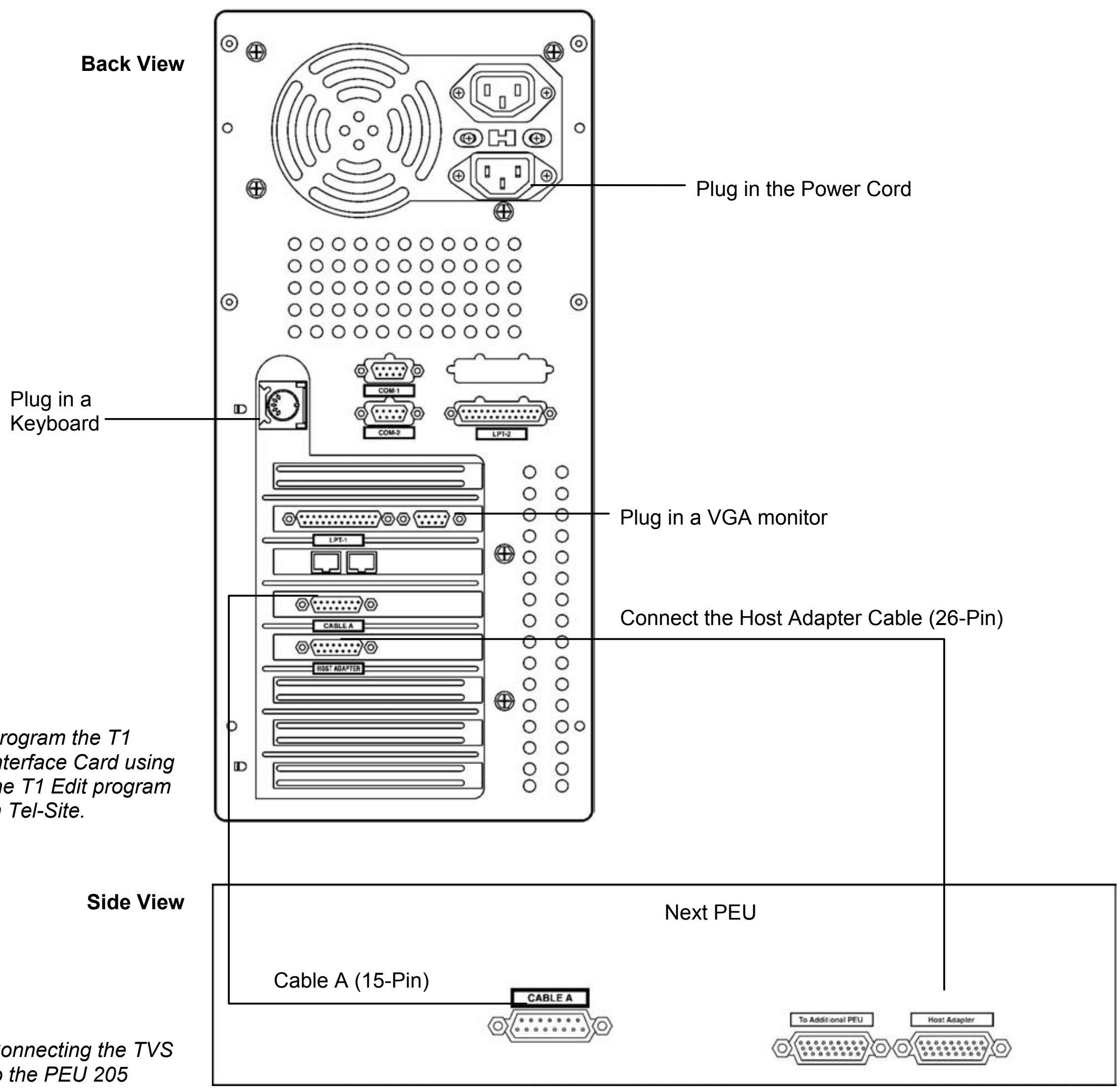
The diagram below shows the correct port for each cable on the TVS and PEU 205. To connect the TVS to the PEU 205, complete the following steps.

1. Plug one end of the 15-pin cable in the Cable A jack on the TVS; plug the other end of the 15-pin cable in the Cable A jack on the PEU.

2. Plug one end of the 26-pin data cable in the Host Adapter port on the TVS; plug the other end of the 26-pin data cable in the Host Adapter port on the PEU.

3. Plug the Power cord into the TVS, and then connect it to a powerstrip.

4. Plug in a keyboard and VGA monitor. This is required to create an emergency boot floppy (page 47).



Connecting the TVS to the PEU 205

Installing Additional PEUs

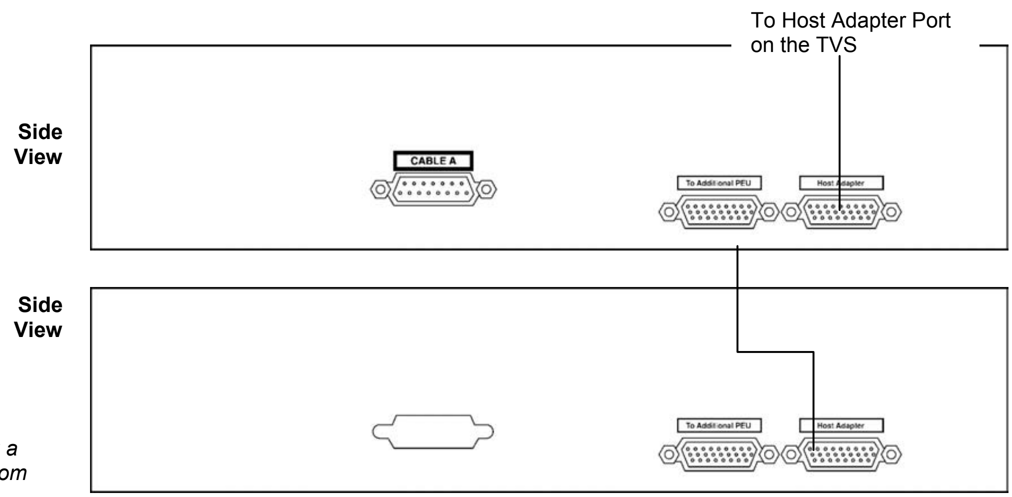
Up to 11 additional PEUs can be added to your VS1 phone system to increase the number of available ports to 192. An additional PEU can be daisy-chained off the first PEU. The first PEU (odd numbered) is always connected directly to the Host Adapter Card inside the TVS, and the second PEU (even numbered) is daisy chained from the first PEU. To add more than two PEUs, you must install additional Host Adapter Cards. *For information on installing a Host Adapter Card, see "Installing a Host Adapter Card," page 35.*

To daisy chain a second PEU off the first PEU, complete the following steps:

1. Connect PEU 1 to the Host Adapter port on the TVS, as described on page 26.

2. Plug a 26-pin data cable in the To Additional PEU port of PEU 1.

3. Plug the other end of the 26-pin data cable in the Host Adapter port on PEU 2.

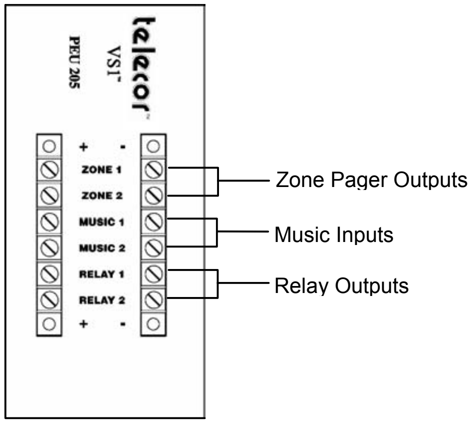


Daisy-chaining a second PEU from the first PEU

If you want to add a third PEU, install an additional Host Adapter Card in the TVS, and then connect the third PEU to the second Host Adapter port on the TVS. Odd-numbered PEUs (PEU 1, 3, 5, 7, 9, 11) must always be connected to a Host Adapter card in the TVS, and even-numbered PEUs (PEU 2, 4, 6, 8, 10, 12) are daisy-chained off the odd-numbered PEUs.

External Contacts – PEU 205

The Port Expansion Unit Model 205 has six external contacts including two zone pager outputs, two music source inputs, and two dry contact relays. The following pages explain how to set up each of the external contacts.



PEU 205 external contacts

Connecting Zone Pager Outputs on a PEU 205

The Zone Pager contacts enable you to connect overhead paging systems to the VS1 phone system. It provides a 600 ohm, low-impedance signal. To connect an overhead paging system to the Telecor VS1 phone system, complete the following steps.

1. Connect a cable (with spade connectors or bare wire) to the “Input” jacks on the paging equipment. The Model 205 accepts spade connectors, number 6 stud, or bare wire. Use 24-gauge wire or larger.

2. Match the negative and positive connectors to the correct jacks on the Model 205.

3. Use a phone connected to the Teleco*r* VS1 phone system to make a test announcement and then adjust the volume level on the paging equipment as necessary.

Connecting Music Inputs on a PEU 205

The music inputs are for audio sources such as radios, music services, tape players, or CD players. They are used for music on-hold and promotions on-hold. Each CO port can be configured for a specific music source. CO ports can also be configured to play digital recordings stored on the TVS hard drive. The music inputs on the Model 205 accept 600 ohm, low-impedance signals. To connect a device to the music inputs on the VS1 phone system, complete the following steps:

1. Use a spade connector, number 6 stud, or bare wire to connect the device to the music source input on the Model 205. Use 24-gauge wire or larger.

   - The cable must be connected to the “Output” jacks on the music source.

2. Connect the ground wire to the negative input terminal on the PEU 205, and then connect the insulated wire to the positive input terminal on the PEU 205. Contact Telecor Technical Support for more information.

Connecting Relay Outputs on a PEU 205

The dry contact relays can be used to attach devices such as electronic door locks, sirens or buzzers. These devices are controlled by dialing a DTMF code from any station. Follow these general guidelines when connecting devices to the dry contact relays.

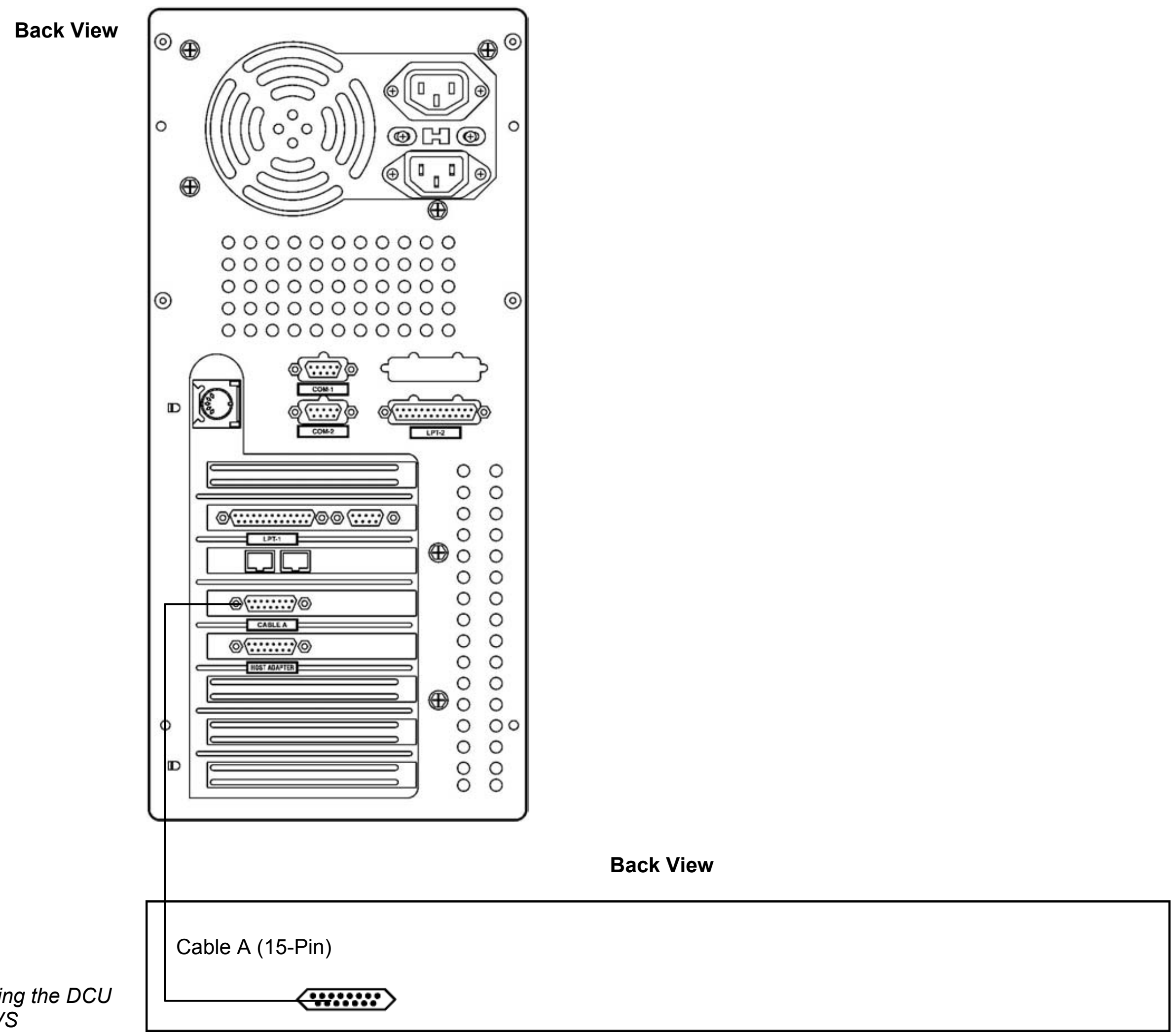
- Each relay output is normally open and completely isolated from the VS1 phone system and from ground.

- The maximum open circuit voltage for either contact is 24 volts AC or DC. The maximum current switched by either contact is 0.5 amps.

Installing the Dry Contact Unit Model 100

To connect the TVS to the DCU, complete the following steps.

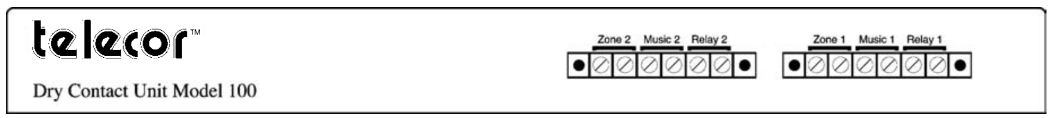
1. Plug one end of the 15-pin cable in the Cable A jack on the TVS.
2.
3. Plug the other end of the 15-pin cable in the Cable A jack on the PEU.



Connecting the DCU to the TVS

External Contacts – Dry Contact Unit Model 100

The Dry Contact Unit Model 100 has six external contacts including two zone pager outputs, two music source inputs, and two dry contact relays. The following pages explain how to set up each of the external contacts.



Model 100 external contacts

Connecting Zone Pager Outputs on a DCU

The Zone Pager contacts enable you to connect overhead paging systems to the VS1 phone system. It provides a 600 ohm, low-impedance signal.

To connect an overhead paging system to the VS1 phone system, complete the following steps.

1. Connect a cable (with spade connectors or bare wire) to the “Input” jacks on the paging equipment. The DCU accepts spade connectors, number 6 stud, or bare wire. Use 24-gauge wire or larger.

2. Match the negative and positive connectors to the correct jacks on the DCU.

3. Use a phone connected to the Telecor VS1 phone system to make a test announcement and then adjust the volume level on the paging equipment as necessary.

Connecting Music Inputs on a DCU

The music inputs are for audio sources such as radios, music services, tape players, or CD players. They are used for music on-hold and promotions on-hold. Each CO port can be configured for a specific music source. CO ports are also configurable to play digital recordings stored on the TVS hard drive. The music inputs on the DCU accept 600 ohm, low-impedance signals.

To connect a device to the music inputs on the VS1 phone system, complete the following steps:

1. Use a spade connector, number 6 stud, or bare wire to connect the device to the music source input on the DCU. Use 24-gauge wire or larger.

    - The cable must be connected to the “Output” jacks on the music source.

2. Connect the ground wire to the negative input terminal on the DCU, and then connect the insulated wire to the positive input terminal on the DCU. Contact Telecor Technical Support for more information.

Connecting Relay Outputs on a DCU

The dry contact relays are used to attach devices such as electronic door locks, sirens, or buzzers. These devices are controlled by dialing a DTMF code from any station. Follow these general guidelines when connecting devices to the dry contact relays.

- Each relay output is normally open and completely isolated from the VS1 phone system and from ground.

- The maximum open circuit voltage for either contact is 24 volts AC or DC. The maximum current switched by either contact is 0.5 amps.

Installation Checklist

- ❑ Configure each PEU by setting the jumper settings for each port.
- ❑ Mount the PEU to the wall or in the 7' x 19" rack.
- ❑ Use the cables to connect the TVS to the first PEU, and then mount the TVS.
- ❑ Connect additional PEUs by installing additional Host Adapter Cards and then daisy chaining them from the first PEU.
- ❑ Connect external contacts

Connecting the Cut-Over Box

When installing a new system, Telecor recommends that the Cut-Over Box cable connections are made after the system has been installed, the software configured, and all stations have been tested and verified. This aids in determining which stations will be connected to the Cut-Over Box. To connect the Cut-Over Box, complete the following steps:

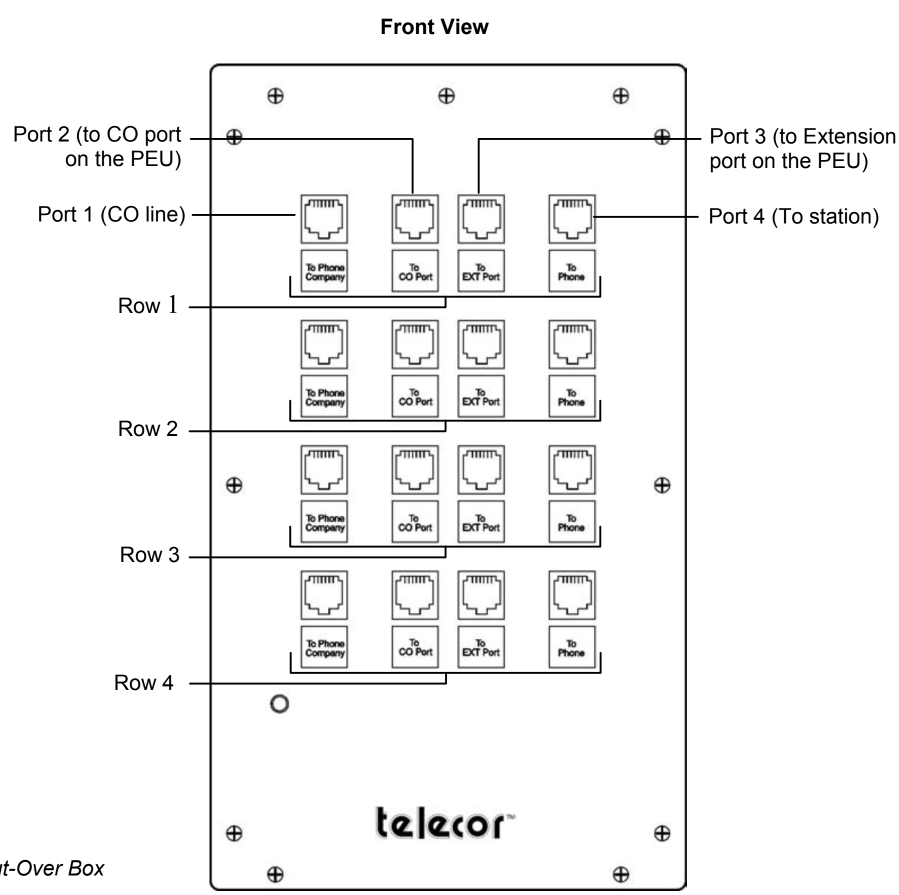
1. Disconnect the RJ-11 cable from the first CO port on the PEU, and then plug it into Port 1 (To Phone Company) on the Cut-Over Box.

2. Use a second RJ-11 Cable to connect the first CO port on the PEU to Port 2 (To CO Port) on the Cut-Over Box.

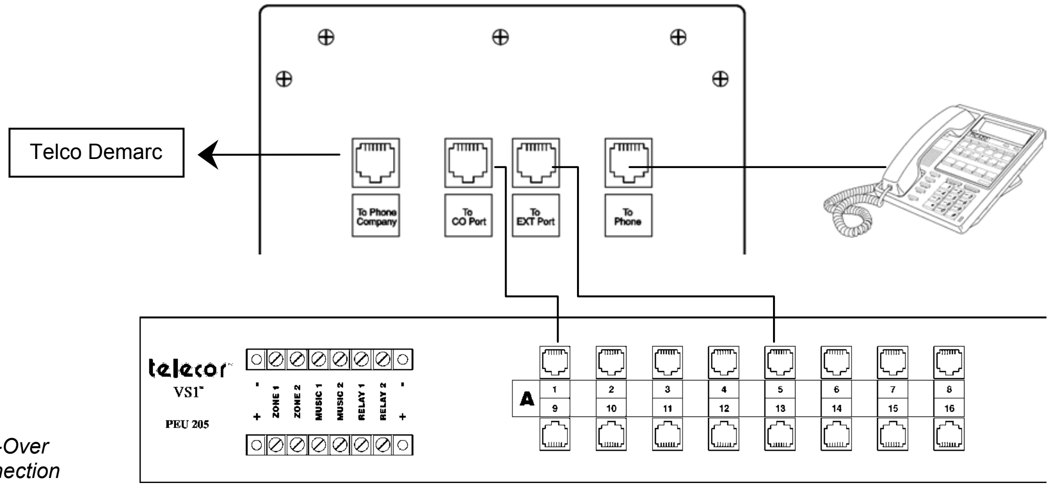
3. Disconnect the RJ-11 cable from the first Station port on the PEU, and then plug it in to Port 4 (To Phone) on the Cut-Over Box.

4. Use a third RJ-11 cable to connect the first Station port on the PEU to Port 3 (To Extension Port) on the Cut-Over Box.

5. Repeat Steps 1-4 for each CO line by using each row on the Cut-Over Box.



Cut-Over Box



Final Cut-Over Box Connection

Testing the Cut-Over Box

After all connections are made, test the Cut-Over Box by turning the switch on the bottom of the Cut-Over Box to the Off position. The power indicator turns off. When the Cut-Over Box is off, calls coming in on a CO line route directly to the corresponding extension.

Note Turning off the Cut-Over Box terminates existing phone calls on the CO lines connected to it. Other lines are not affected.

Note If a power outage occurs, the Cut-Over Box automatically switches the connected CO lines to their dedicated extensions.

Cut-Over Box Checklist

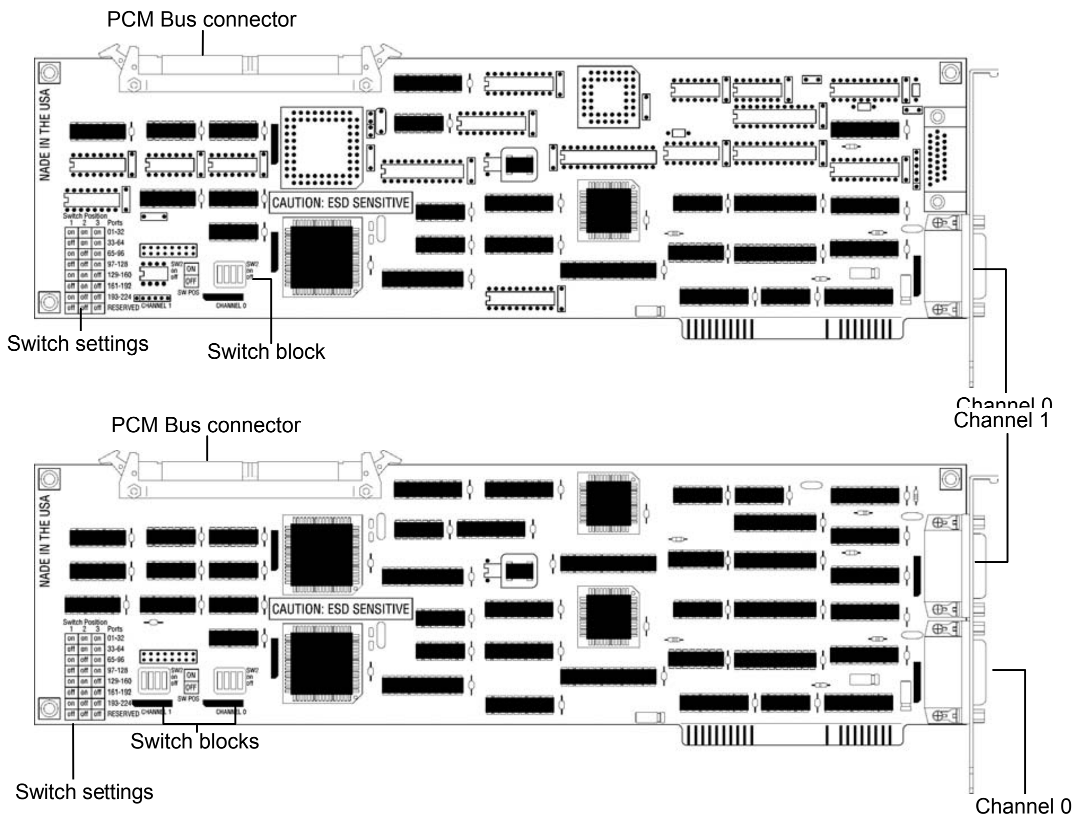
- ❑ Mount the Cut-Over Box in the rack or to the wall.
- ❑ Disconnect the RJ-11 cable from the first CO port on the PEU, and then connect it to Port 1 on the Cut-Over Box.
- ❑ Connect the first CO port on the PEU to Port 2 on the Cut-Over Box.
- ❑ Disconnect the RJ-11 cable from the first station port on the PEU, and then plug it into Port 4 on the Cut-Over Box.
- ❑ Connect the first station port on the PEU to Port 3 on the Cut-Over Box.
- ❑ Connect additional CO ports to the other rows on the Cut-Over Box.
- ❑ Turn off the Cut-Over Box to test it

Installing a Host Adapter Card

Each TVS comes with one 32-port Host Adapter Card installed. Additional Host Adapter Cards (either 32-port or 64-port models) are inserted into any unused ISA slot. Installing three 64-port Host Adapter Cards expands the number of ports to 192. An Internal PCM Bus connector is attached to the top of the Host Adapter Card, and is attached to any additional Host Adapter Cards already installed in the TVS.

Each Host Adapter Card has switches that must be set to tell it which port numbers to support. A switch settings table is printed on the Host Adapter Card for reference during installation.

Note	Each Host Adapter Card has a unique Serial Number. The TVS uses the Serial Number of the first Host Adapter Card (ports 1-32) as the System ID Number. The System ID Number is used to authenticate Activation Keys for that particular TVS. Replacing the Host Adapter Card configured for ports 1-32 requires new Activation Keys for that TVS.



32-Port Host Adapter Card

64-Port Host Adapter Card

To install a Host Adapter Card in the TVS, complete the following steps:

1. Set the port switches on the Host Adapter Card to correspond with the PEU ports it is going to support. Use the following table, or refer to the one printed on the Host Adapter Card. 64-port Host Adapter Cards have two switch blocks to set.

Set to Ports	SW1	SW2	SW3	SW4
1-32	ON	ON	ON	OFF
33-64	OFF	ON	ON	OFF
65-96	ON	OFF	ON	OFF
97-128	OFF	OFF	ON	OFF
129-160	ON	ON	OFF	OFF
161-192	OFF	ON	OFF	OFF

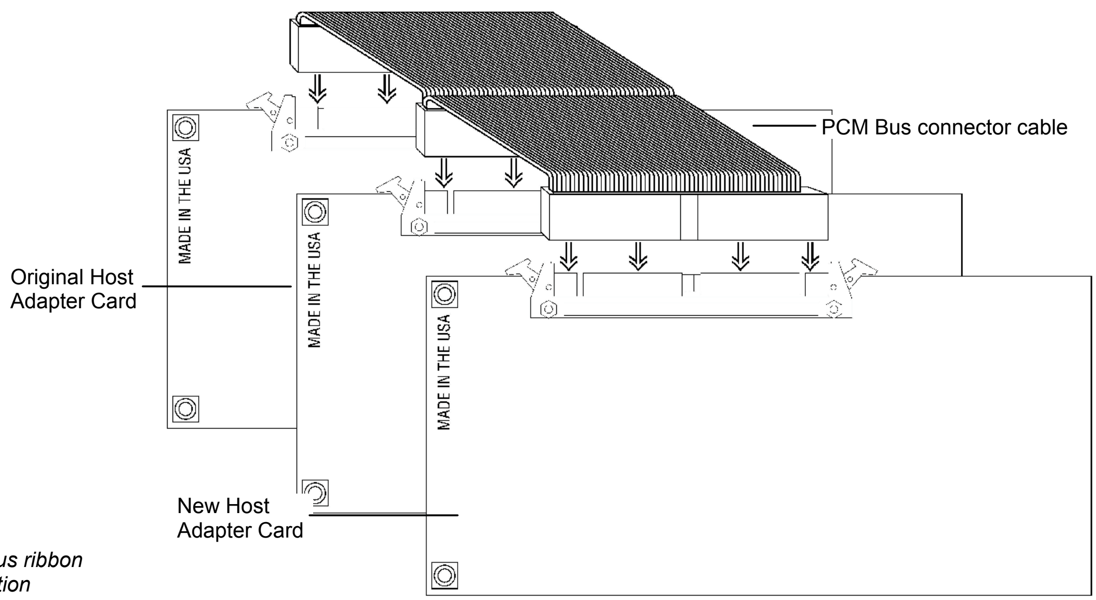
Note Each Host Adapter Card must be configured for the range of ports it will support. Do not configure two Host Adapter Cards for the same range of ports, or the system will not work.

2. Disconnect the CO lines from the PEU, and then turn off the system and unplug it. Disconnect the Host Adapter Cable. Remove the TVS cover, and then lay the TVS on its side with the front facing forward.

3. Locate an available ISA slot, and then remove its slot cover from the back of the TVS case.

4. Insert the Host Adapter Card into the selected ISA slot, and then secure it with the screw from the slot cover.

5. Connect the new Host Adapter Card to the PCM Bus connector cable already attached to the first Host Adapter Card and the Switch Card.



PCM Bus ribbon connection

6. Replace the TVS cover, plug in the system, and then turn the power on. Reconnect the CO lines to the PEU.

After installing a Host Adapter Card, you can connect additional PEUs to your *Telecor* VS1 phone system. *For more information, see "Installing Additional PEUs," page 28.*

Host Adapter Cards Checklist

- ❑ Set the port switches on the Host Adapter Card.
- ❑ Disconnect CO lines from the PEU, turn off the system, and then remove the TVS cover.
- ❑ Locate an available ISA slot, and then remove its slot cover.
- ❑ Install the Host Adapter Card in the TVS.
- ❑ Connect the Host Adapter Card to the PCM Bus cable in the TVS.
- ❑ Replace the TVS cover and reconnect the Host Adapter cable. Plug in and turn on the system, and then reconnect CO lines to the PEU.
- ❑ Connect additional PEUs to the TVS.

Installing the Caller ID Option Module

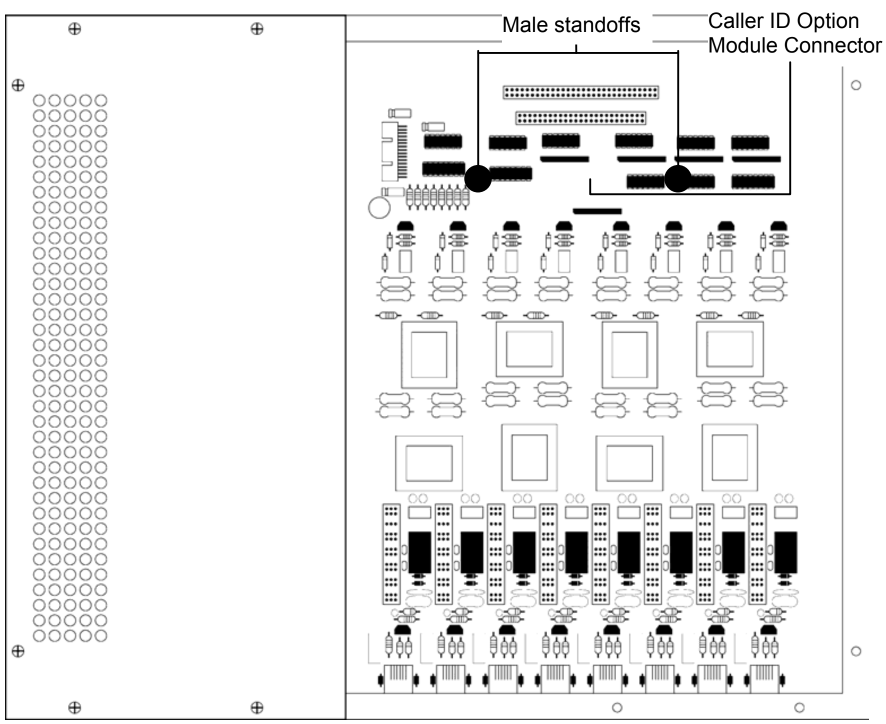
The Caller ID Option Module is installed inside a Port Expansion Unit Model 200 (PV-PEU-200) or Port Expansion Unit Model 205 (PV-PEU-205), serial number 40,000 and above. The Caller ID module is not used with the Port Expansion Unit Model 250 (PV-PEU-250) as it contains integrated Caller ID circuitry on all 16 ports. If you want to set up Caller ID on CO lines connected to additional PEUs, you must install additional Caller ID Option Modules in those PEUs.

To install the Caller ID Option Module inside a PEU, complete the following steps:

Warning! Before installing the Caller ID Option Module, first touch the metal case of the PEU to discharge any static electricity. Static electricity can damage the Caller ID Option Module and your system.

1. Disconnect the CO lines from the PEU, and then turn off the system. Unplug the PEU and remove its cover.

2. Find the two male standoffs in the PEU. They are located on either side of the 50-pin connector.



Male Standoffs in the PEU

Note Some PEUs have female standoffs instead of male standoffs. To install the Caller ID Option Module on a PEU with female standoffs, place the holes on the Module over the female standoffs on the PEU. Use the two screws provided, and place them into the holes to secure the Caller ID Option Module. If the Module is not properly installed, the holes do not align and the screws do not fit.

3. Install the Caller ID Option Module with the component side facing down. Align the two holes on the Module with the male standoffs.

4. Use the two nuts provided, and fasten them onto the standoffs to secure the Caller ID Option Module. If the Module is not properly installed, the holes do not align, and the nuts do not fit on the standoffs.

Warning!	If the Caller ID Option Module is not installed correctly it will damage your equipment. Improper installation voids the warranty on the Caller ID Option Module and on the PEU into which it is installed.

5. Replace the cover on the PEU, and then plug it in.

6. Reconnect the CO lines.

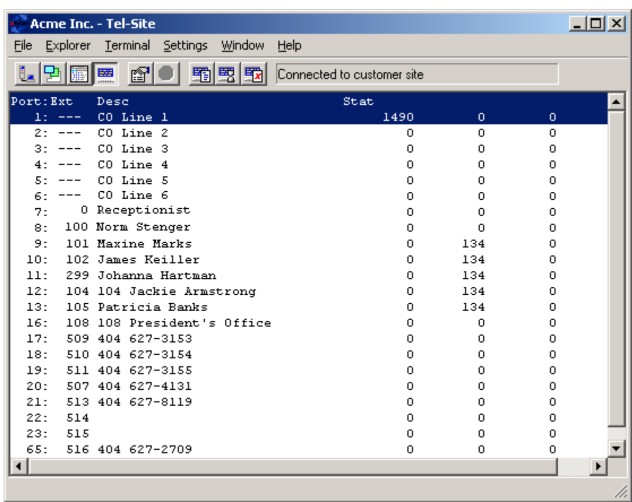
See "Port Configurations" in the VS1 Editor section for activating Caller ID.

Verifying the Caller ID Option Module

When the Caller ID Module is properly installed and configured, Caller ID numbers are sent to station option displays. If a CID Error on the station option display is received, the system could not find the Caller ID Module board. If an N/A error on the station option display is received, the system could not find the Caller ID service from the phone company. Contact the phone company to ensure that Caller ID service has been activated.

To verify the Caller ID Option Module:

1. Using the **Terminal** window in the Tel-Site system management application, type **`ls`** at the TVS Command prompt. This displays a list of all ports and their status.
2. Press the SPACEBAR to open the **Polling** window in the **Line Status** window. The first column represents a device polled and responding correctly; the second column represents a device polled with no response; the third column represents a device polled with an invalid response. To restart the counters, type **`r`**.
3. If the Caller ID Module is working, the first CO Line port shows activity in the first column. If there is activity in the second column, the Caller ID Module is not functioning properly.



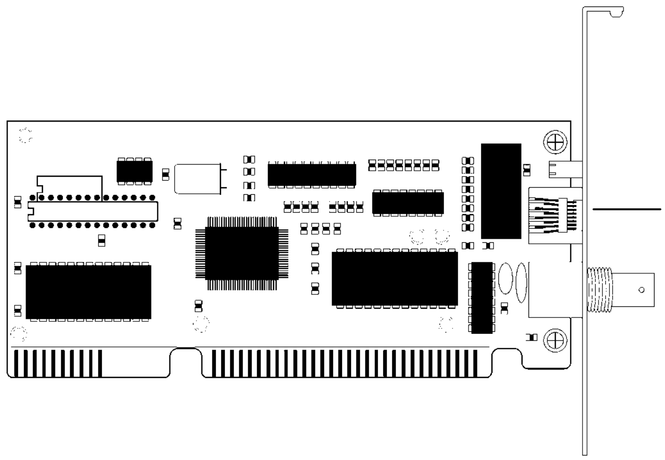
The Polling Window

Caller ID Option Module Checklist

- ❑ Disconnect the CO lines from the PEU, and then unplug it.
- ❑ Install the Caller ID Option Module in the PEU.
- ❑ Confirm that Caller ID has been activated by the phone company.
- ❑ Use the VS1 Editor configuration program to activate Caller ID.
- ❑ Verify that Caller ID is working.

Installing a 10Base-T Network Interface Card

The 10Base-T Network Interface Card is inserted into any unused 16-bit ISA slot. It has a preconfigured I/O address of 340 IRQ 11. An RJ-45 connector is provided on the 10Base-T Network Interface Card for connection to the network.



10Base-T Network Interface Card

To install a 10Base-T Network Interface Card in the TVS, complete the following steps:

1. Disconnect the CO lines from the PEU, and then turn off the system and unplug it. Remove the TVS cover, and then lay the TVS on its side with the front facing forward.

2. Locate an available 16-bit ISA slot, and then remove its slot cover from the back of the TVS case.

3. Insert the 10Base-T Network Interface Card into the selected ISA slot, and then secure it with the screw from the slot cover.

4. Replace the TVS cover, and then connect the RJ-45 cable from the Network to the Connector on the 10Base-T Network Interface Card.

5. Plug in the system, and then turn the power on. Reconnect the CO lines to the PEU.

Configuring the 10Base-T Network Interface Card

The TVS automatically detects the presence of the 10Base-T Network Interface Card. Complete the following steps to connect to the network.

Note	If your system uses switch card model 100 and a network card configured for use with IRQ11, the TVS will not detect the 10Base-T Interface card. Call Telecor Technical Support for assistance.

1. Using the VS1 Editor configuration program, select the **System Configuration 2** pane.

2. Enter the following information in the **Network** group box:

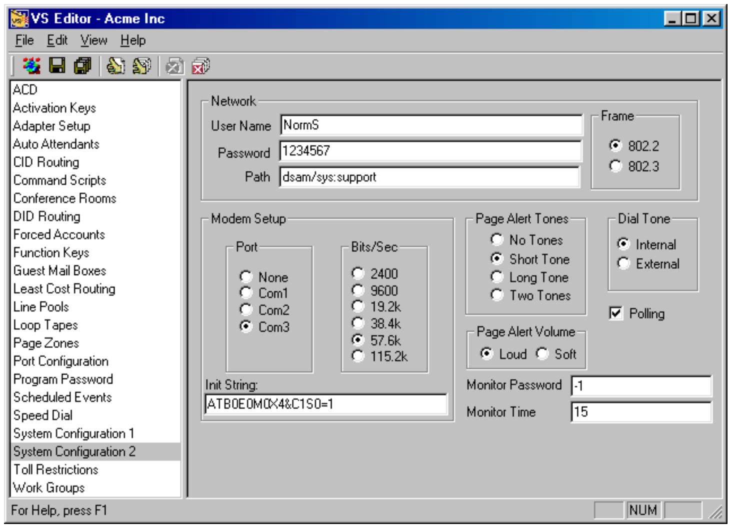
UserName: Type a valid UserName on the network. The UserName must have a password to work correctly. The UserNameis a case-sensitive box.

Password: Type a valid password on the server for the user described in the **UserName** box. Password is a case-sensitive box.

Path: Path is the server name, volume, and path that you want the SMDR information written to on your server. For example, **`dsam/sys:support`** configures the network path to the support directory on the **`sys`** volume of the DSAM server.

3. In the **Frame** group box, set the frame to match the frame type of the network server that you are logging on to.

System Configuration 2 pane



4. Click the Save button in the toolbar to save your changes

See "Station Message Detail Recording (SMDR)" in the Reference section to redirect SMDR files through a network.

10Base-T Network Interface Card Checklist

- ❑ Disconnect the CO lines from the PEU, turn off the system, and then remove the TVS cover.
- ❑ Locate an available 16-bit ISA slot, and then remove its slot cover.
- ❑ Install the 10Base-T Network Interface Card in the TVS.
- ❑ Replace the TVS cover, and then connect the RJ-45 cable from the Network to the Connector on the 10Base-T Network Interface Card.
- ❑ Plug in and turn on the system, and then reconnect the CO lines to the PEU.
- ❑ Use the VS1 Editor configuration program to configure the 10Base-T Network Interface Card.
- ❑ See "Station Message Detail Recording (SMDR)" in the Reference section to redirect SMDR files through a network.

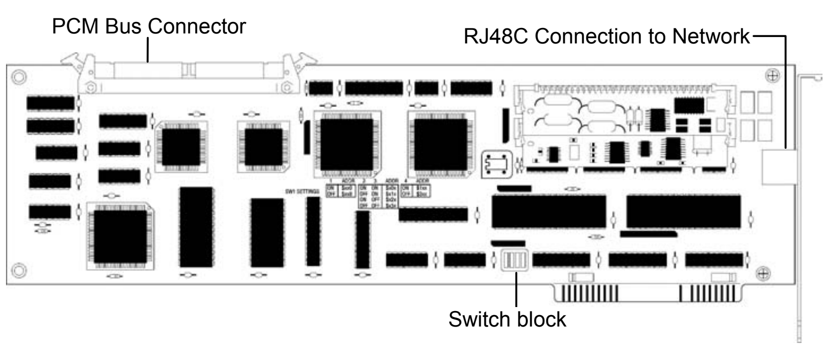
Installing a T1 Interface Card

The T1 Interface Card is installed inside the Telecor Voice Server (TVS) and supports 24 T1 channels. A maximum of two T1 Interface Cards can be installed in the TVS. The following table shows the number of T1 channels and Analog CO lines that can be configured for each T1 card installed in the TVS.

Number of T1 Cards	T1 channels	Maximum # of PEUs	Maximum Analog CO lines	Maximum Total Ports
0	0	12	96	192
1	24	10	72	184
2	48	8	48	176

The T1 Interface Card is an ISA card and is inserted into any unused ISA slot. The internal bus connector fits on top of the T1 Interface Card that attaches it to the other Telecor VS1 phone system cards. The T1 Interface Card has a built-in channel service unit, and no additional Telecor VS1 phone system hardware is required. Switch settings for T1 Interface Cards are different from the switch settings used for Host Adapter Cards. The first T1 Interface Card (T1 Card 0) must be installed into the range of ports from 97 to 128. The second T1 Interface Card (T1 Card 1) must be installed into the range of ports from 65 to 96.

Note	If you already have a Host Adapter Card installed and a PEU is configured for Ports 96–128 or Ports 65–96, change the PEU port assignments before configuring the T1 Interface Card.



T1 Interface Card

Ordering T1 Service

Before installing a T1 Interface Card, determine what type of service is required on-site, and then order T1 service from the local phone company. When placing an order, provide the FCC ID number from the label on the T1 Interface Card. You have the following T1 service options:

- **DID (Direct Inward Dialing)**—reports the number that was dialed by the calling party. A Service Provider determines the number of digits in the number string and assign a block of numbers to a DID hunt group. The smallest block is usually 100 numbers. These numbers can be used as direct dials to extensions, Guest Mailboxes or Voice Mail for an extension.

- **ANI (Automatic Number Identification**)—reports the number of the calling party. This is similar to Caller ID on an analog line. However, the only information provided is the number from which the caller is calling. Name information is not sent by the Service Provider.

- **DNIS (Dialed Number Identification Service)**—reports the number that was dialed by the calling party. A Service Provider determines the number of digits in the number string

(minimum=1, maximum=10) and assigns numbers as you want. For example, you may want only the last four digits of similar numbers in order to differentiate between them. DNIS is typically only available on inbound 800 service.

Outbound lines may or may not have a phone number assigned. If a phone number is assigned by the Service Provider, these lines can be used to receive incoming calls.

The T1 Interface Card receives DID or ANI information using MF (Multi-Frequency) or DTMF (Dual Tone Multi-Frequency) tones. The T1 Interface Card automatically determines if MF or DTMF is present on a call-by-call basis. Dial pulses are not generated or used by the T1 Interface Card.

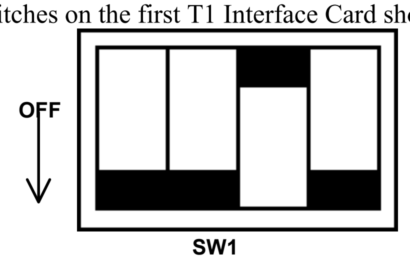
When ordering T1 service, request a framing mode of ESF (Extended Super Frame), but be aware that many Service Providers supply only D4 framing. ESF provides troubleshooting capabilities that are not available with D4. If ESF framing mode is available, request B8ZS (Bit Eight Zero Suppression) line coding. If ESF is not available, AMI (Alternative Mark Inversion) line coding is usually used.

Installing the T1 Interface Card

To install the T1 Interface Card, complete the following steps:

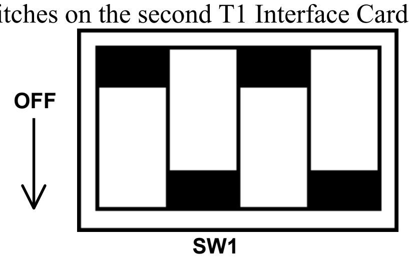
1. Verify that the switches on the T1 Interface Card are set to the correct positions.

   - Switches on the first T1 Interface Card should be set for ports 97–128.



Switch settings for
T1 Card 0
Ports 97-128

   - Switches on the second T1 Interface Card should be set for ports 65–96.



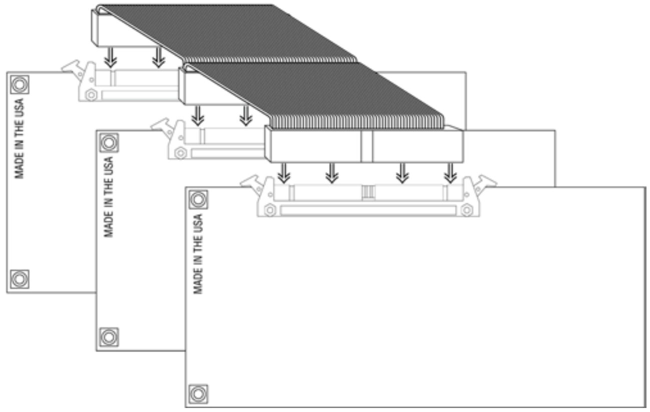
Switch settings for
T1 Card 1
Ports 65-96

2. Disconnect the CO lines from the PEU and the Host Adapter cables connecting the TVS to the PEU. Turn off the TVS, unplug it, and then remove the cover.

3. Locate an available ISA slot, and then unscrew and remove its slot cover from the back of the TVS case.

4. Insert the T1 Interface Card in the selected ISA slot. Push firmly to set the T1 Interface Card into the slot and to ensure proper connection, and then secure it with the screw from the slot cover.

5. Connect the PCM bus cable to the connector at the top of the card.

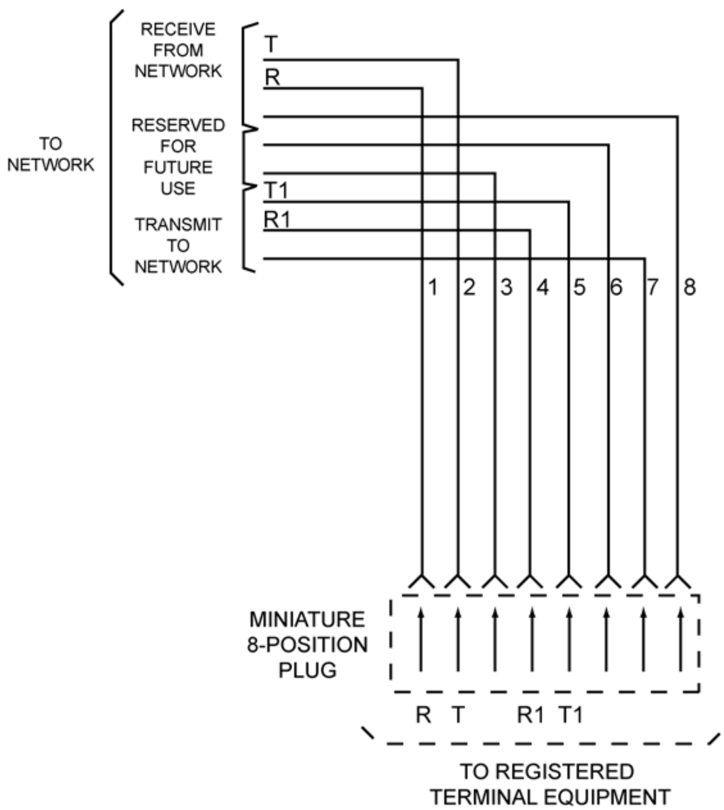


Connecting the T1 Interface Card to the ribbon bus cable

6. Replace the TVS cover, and then reconnect the Host Adapter Cables. Turn the system on, and then reconnect the CO lines.

Cable Installation

After installing the T1 Interface Card, make the cable connection from the Telco Demarc (T1 repeater) point to the RJ48C jack on the end of the T1 Interface Card. Telecor provides a cable with connectors. If a longer cable is required, it must be constructed from Category 5 twisted pair wiring connected to pairs 1 and 2 with an RJ48C connector at each end.



T1 cable wiring diagram

See *"T1 Interface Card Configuration" in the Reference section for setting up the T1 Interface Card.*

Installing the CTIM

The Computer Telephony Interface Module (CTIM) provides a digital interface to the VS1 phone system for the Telecor Attendant and Connect applications. The CTIM is an external device with a headset jack intended for use only with the VS1 phone system. To install the CTIM, complete the following steps:

1. Close all applications, and then turn off the computer.

2. Connect the female end of a 9-pin serial port cable to an available 9-pin male COM port on the computer you want to use with the CTIM.

Note If you computer has an available 25-pin COM port, obtain a 25-to-9-pin serial port adapter and connect that to the 25-pin COM port on the computer.

3. Connect the male end of the 9-pin serial cable to the DB9 female connector on the CTIM.

Note If you are installing the CTIM for use with Telecor Attendant, the DB9 connector on the CTIM is not used.

4. Connect a four-conductor RJ-11 patch cord from the wall jack to the RJ-11 jack labeled "Line" on the CTIM.

5. Connect a headset to the RJ-22 jack labeled "Headset" on the CTIM.

6. Set the Ear, Mic and Sidetone volume control jumper settings on the CTIM for the headset or handset you use. The table below shows the volume settings for Ear and Mic.

7. Set the Sidetone to Hi or Lo, according to user preference. The Sidetone is set to Low by default.

8. The CTIM installation is complete. Turn the computer on.

CTIM
Computer Telephony Interface Module

Volume settings for Ear and Mic

Volume	E2/M2	E1/M1	E0/M0
1(LO)	off	off	off
2	off	off	on
3	off	on	off
4	off	on	on
5	on	off	off
6	on	off	on
7	on	on	off
8 (HI)	on	on	on

CTIM Checklist

- ❑ Turn off the computer
- ❑ Locate an available external 9-pin male COM port on the computer, and connect the female end of a serial cable to it.
- ❑ Connect the male end of the serial cable to the female connector on the CTIM.
- ❑ Connect an RJ-11 cord from the wall jack to the "Line" jack on the CTIM. Connect a headset to the RJ-22 "Headset" jack on the CTIM.
- ❑ Set the Ear, Mic, and Sidetone volume control jumper settings on the CTIM.
- ❑ Turn the computer on.

Installing the PC Option Module (PCOM)

The PC Option Module (PCOM) provides an externally connected serial interface to a computer for data sent by the Telecor Voice Server (TVS). The PCOM is used with the Telecor Connect CTI client application. By installing a PCOM, Connect users can use their computer screen for call processing and keep the phone on their desk. Keeping a phone on the desk also allows calls to be received when the computer is turned off. To install the PCOM, complete the following steps.

1. Close all applications, and then turn off the computer. Locate an available 9-pin or 25-pin serial port on the back of the computer.

2. If there is not an available serial port on the computer, you need to install one.

3. Connect the 9-pin DB9 female connector of the PCOM cable adapter to the 9-pin DB9 male connector of the computer.

4. Connect the phone line from the station jack to one of the RJ-11 jacks on the PCOM.

5. Connect the phone to the other RJ-11 jack of the PCOM. PCOM installation is complete. Turn the computer on.

PCOM Checklist

- ❑ Turn off the computer
- ❑ Locate an available external 9-pin or 25-pin serial port or install one.
- ❑ Connect the PCOM to the computer using the 9-pin DB9F connectors.
- ❑ Connect the phone line and phone to the RJ-11 jacks on the PCOM.
- ❑ Turn the computer on.

CREATING AN EMERGENCY BOOT FLOPPY

An Emergency Boot Floppy must be created to restart the TVS in case of system failure. The disk is created on-site with a monitor and keyboard connected to the TVS. To create an Emergency Boot Floppy you need one ***high-density, write-enabled*** disk. The procedure takes approximately 15 minutes and must occur during downtime. Complete the following steps:

1. If the server is running with the Command prompt displayed (**Command - >**), type **stop** or **exit** and then press ENTER on the keyboard.

   or

2. If the server is not running and the DOS prompt is displayed (**C:\>**), press the RESET button the server.

   - The server restarts.

3. When the **VS1 Options** menu appears, press **2** to select the **VS1 Configuration** option.

4. On the **VS1 Configuration** menu, press **4** to select the **Create an Emergency Boot Floppy** option.

5. Insert the high-density, write-enabled disk into Drive A of the TVS, and then press ENTER.

   - The TVS formats the disk, then copies the necessary files to it. In addition to copying the files needed to boot the TVS, a copy of the system configurations is saved to the disk. This procedure takes several minutes to complete.

6. Remove the disk when prompted. Label the disk and store it in a safe place.

7. Press any key when the process is complete.

Note	Voice Mail and SMDR data cannot be saved while the system is operating from an Emergency Boot Floppy.

8. When the **VS1 Configuration** menu (or TVS Startup menu) appears, press **0** to restart the server.

Tel-Site

OVERVIEW OF TEL-SITE

Tel-Site provides remote system access (RSA) to the VS1 phone system. Tel-Site is used for the following:

- Connecting with the customer TVS to upload configuration changes
- Downloading and uploading customer TVS files
- Performing file maintenance
- Diagnosing problems
- Setting the date and time on the customer TVS
- Setting up T1 parameters (for systems with a T1 card)

This section first introduces the windows that Tel-Site uses and how to set up a new customer site. It then documents the different connection methods available for connecting to a site. Finally, it provides basic Tel-Site operating instructions, which cover the following:

1. Connecting to a site.
2. Downloading any changed files from a site.
3. Accessing the VS1 Editor to change site configurations.
4. Uploading files to a site.
5. Reloading Configuration changes.

The Tel-Site application is available for download from the VS1 Dealer Page of the Telecor Web Site (www.telecor.com). Please visit the page for details on obtaining licensing information and installing the application.

The Tel-Site application is a Windows®-based program that operates on Microsoft® Windows® 98, Windows® 2000, and Windows® XP operating systems.

To work properly, the modems used for Tel-Site should be 14.4 kilobits per second (Kbps) or faster.

Tel-Site Windows

Tel-Site includes the following windows:

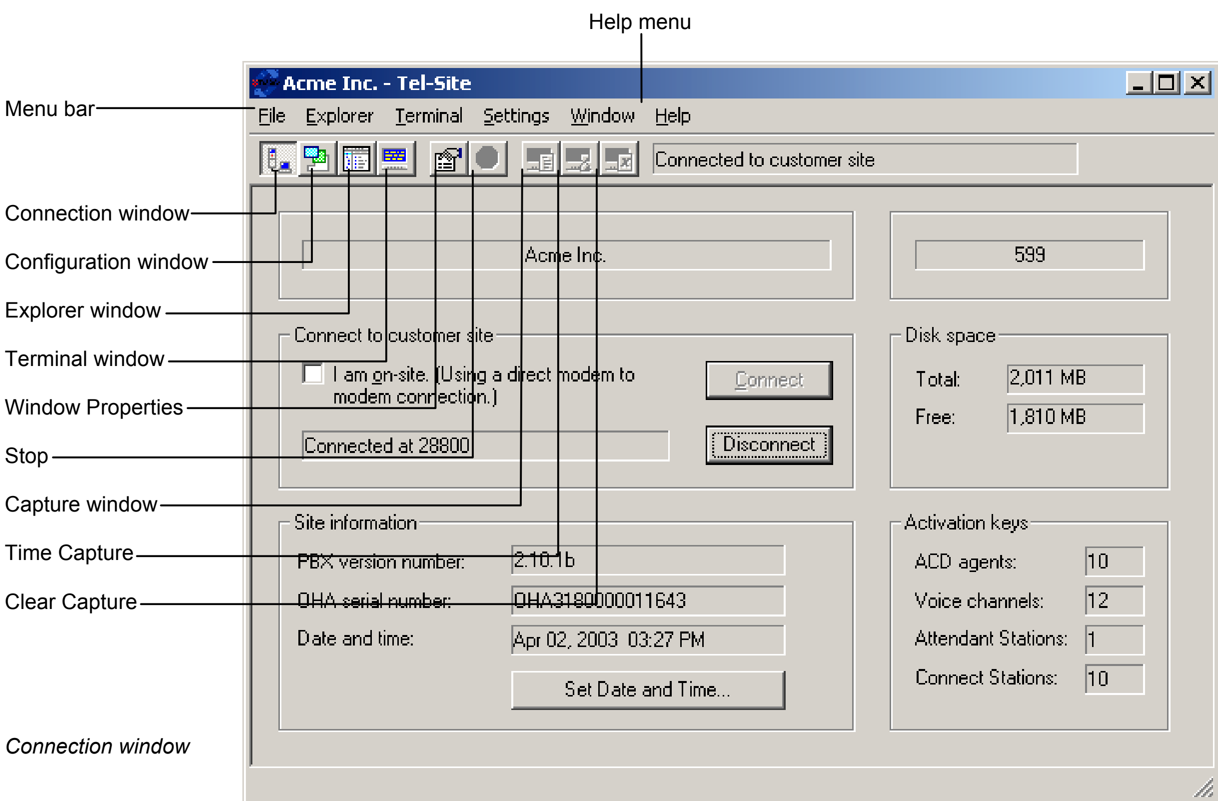
Connection Window – to connect or disconnect from a site
Configuration Window – to configure a customer site
Explorer Window – to perform discretionary file maintenance
Terminal Window – to provide a snapshot of the real-time dynamics of a site for troubleshooting purposes.

The following pages include brief descriptions of each window. For further information, see the Tel-Site Help file.

The Connection Window

The **Connection** window for Tel-Site appears when the application is first started. After setting up a customer site profile (see *"Setting up a New Site," page 56)*, the **Connection** window is where you can connect or disconnect from a site. In the **Connect to customer site** group box there is a **Connect** and **Disconnect** command button.

Once connected, information about the Telecor Voice Server at the site is displayed. This includes site information, such as the version number of the Voice Server at the site, information about free space on the hard disk, and activation keys. You can also change the date and time of the Voice Server.



Connection window

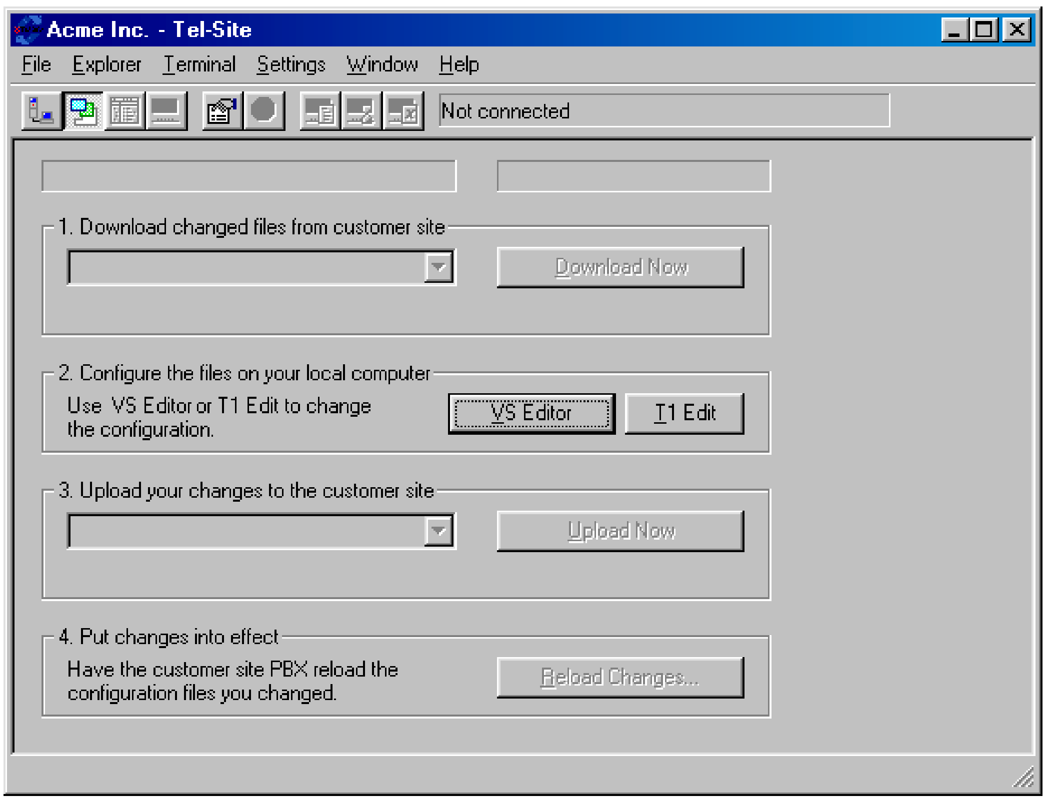
The Configuration Window

The **Configuration** window is used to perform remote configuration of a customer site. When connected to a customer site and the **Configuration** window is opened, Tel-Site scans and displays any TVS files that have changed since the last remote activity. It can then download any updated files so that your local computer matches the latest customer site configuration. The **VS1 Editor** configuration program is then used to configure the files on your local computer. Once configured, the changes are uploaded to the customer TVS and put into effect.

The **Configuration** window is also used to configure a T1 Interface Card by clicking **T1 Edit**. *See "T1 Interface Card Parameter Setup Using Tel-Site" in the Reference section for more information.*

Note Typically, the **Connection** window is used to connect to a customer site, but the Configuration window can be opened prior to connecting to a site to examine the last downloaded configuration.

Note The first time you open the **Configuration** window after upgrading the TVS software, a dialog box opens indicating an upgrade has occurred. You must download all of the Configuration files so they correspond with the latest VS1 software version.



Configuration window

The Explorer Window

The **Explorer** window is designed to perform discretionary file maintenance. It can be used to upload and download files that are not automatically handled by the **Configuration** window.

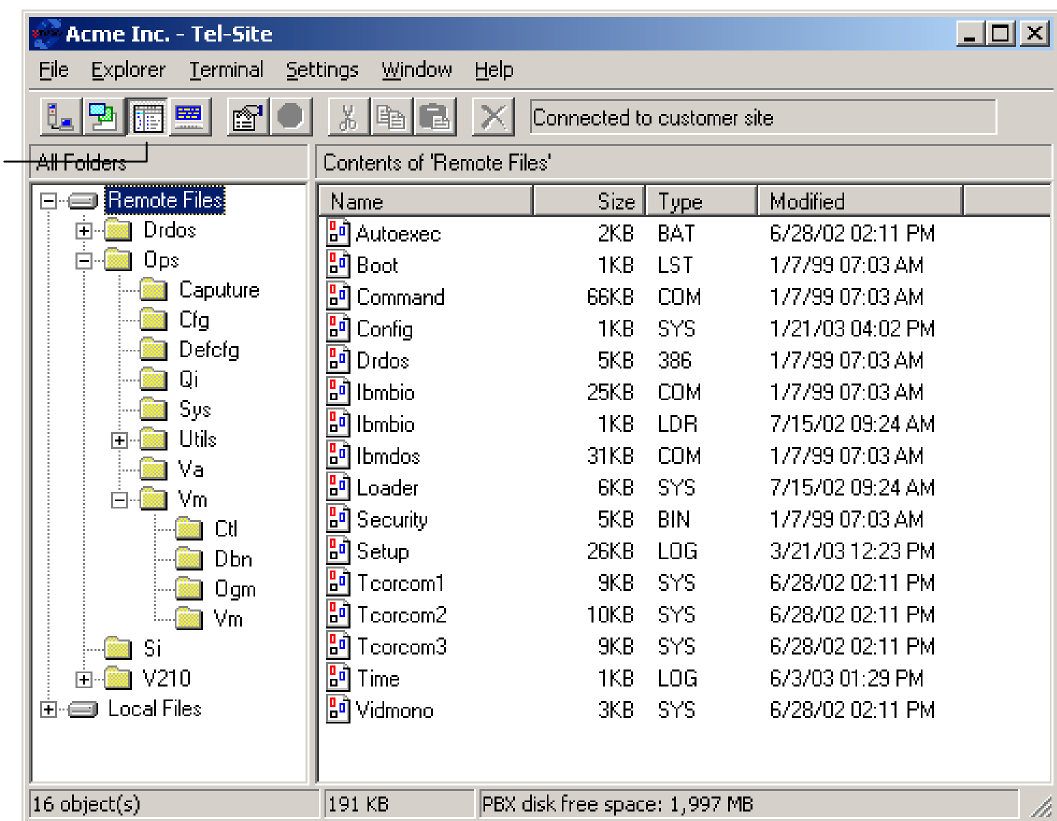
The Tel-Site application keeps track of all changes made to the customer TVS files within the **Explorer** window. The **Explorer** window does not keep track of changes made in the **Connection** window or the **Terminal** window. The status panel at the bottom of the **Explorer** window shows the amount of free disk space available on the TVS.

`The **Explorer** window design is similar to the Windows® Explorer® and many of the features are the same. However, you cannot place or delete files in the Recycle Bin. In addition, the Tel-Site application considers certain folders and files on the customer TVS as critical to operation. These critical files and folders cannot be deleted.

WARNING

WARNING! Because you can upload any file to the TVS, there is a risk of uploading a damaged or incompatible file if you do not have a complete knowledge of the files required for TVS operation. Uploading a damaged or incompatible file can be as bad as deleting the file.

Explorer window toolbar button

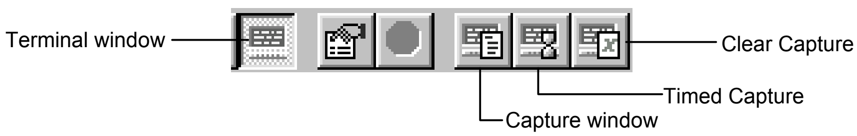


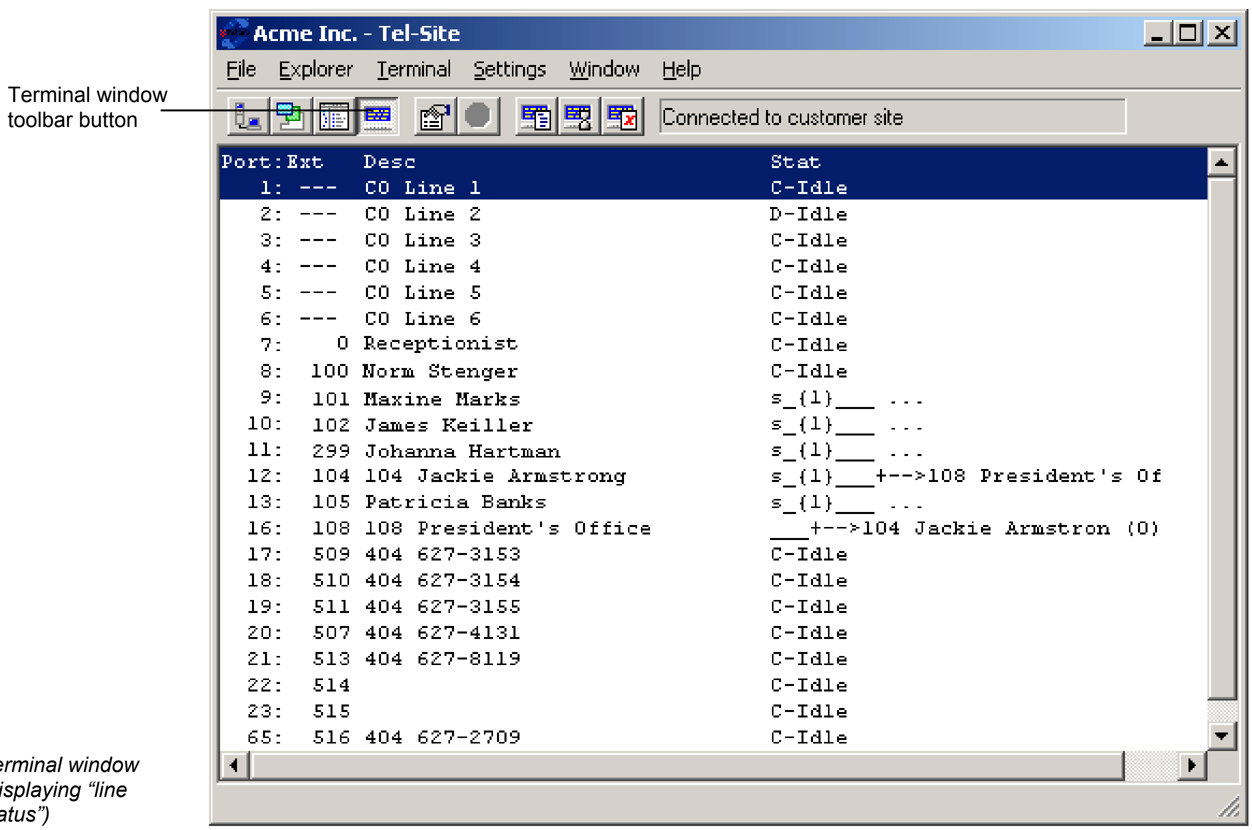
Explorer window

The Terminal Window

The **Terminal** window simulates the monitor display of the TVS for troubleshooting a site. It provides snapshots of the real-time dynamics of a remote phone system. With the **Terminal** window you can capture and create a text file of important configuration information from a remote site. For example, you can capture a T1 setup and create a text file for later reference (if the file does not exist, it is created automatically).

The toolbar has four buttons used for the **Terminal** window: **Terminal Window**, **Capture Window**, **Timed Capture**, and **Clear Capture** buttons. The latter three buttons appear dimmed, or unavailable, until the **Terminal** window is activated.





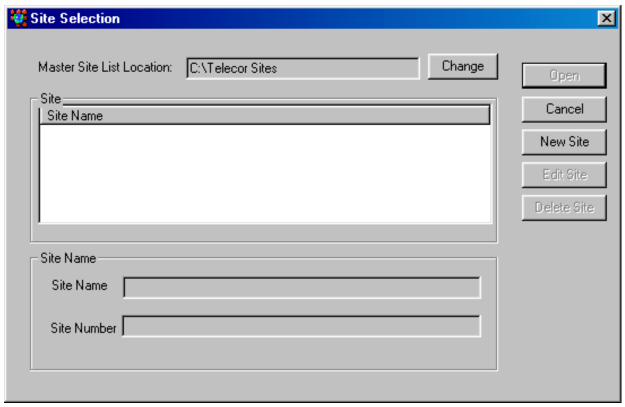
Terminal window (displaying "line status")

Setting up a New Site

To set up a new customer site, complete the following steps:

1. From the **File** menu, click **Site Selection**.

    - The **Site Selection** window appears.



Site Selection window

2. By default, Site Selection will store customer sites in the **C:\Telecor Sites** directory, as displayed in the **Master Site List Location** box. To change the Master Site List Location, click **Change** and navigate to the directory where sites will be stored.

3. Click **New Site**.

    - The **New Customer Site** dialog box appears.

4. In the **Customer Site Name** text box, enter a name or description for the customer site.

5. Click **Next**.

6. Leave the **Phone Number** text box blank. This box will be filled in later when you connect to the TVS for the first time.

7. Leave the **RSA password** text box blank. If the TVS is later assigned a Remote System Access password, then you will be instructed to fill this box in. *See "Remote System Access Password" on page 78 for more information.*

8. Uncheck the **Modem shares a CO line** check box.

9. Click **Finish**.

10. The **Site Selection** window appears with the new site listed under the **Site Name** column.

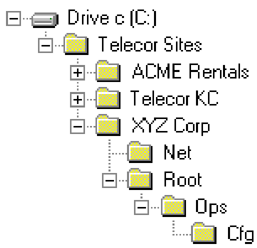
Notes:

- Site information can be changed by clicking **Edit Site** in the **Site Selection** window.

- The **New Customer Site** dialog box is available only when you are not connected to a customer site.

How Tel-Site Organizes Site Information

Tel-Site makes it easy for you to organize multiple customer sites within a single folder on your local computer. By default, this folder is **C:\Telecor Sites**, but you can use a different folder if you like. The illustration shows three customer sites arranged this way.



The XYZ Corp folder also shows the folders that map to the customer TVS. The Net folder represents the network drive on the TVS, if present. The Root folder represents the TVS root folder, and the **Ops** and **Cfg** folders map directly to their counterparts on the TVS.

When you set up a new customer site, you should choose a descriptive name for the site. This name serves both as the site name and the folder name, such as XYZ Corp in the illustration. (If you choose to keep customer sites on a drive that does not support long file names, you will need to use a short name for the customer site. You can later change the site name to something more descriptive if you like.)

Tel-Site does not require that the customer site name and the folder name remain the same. You can change the site name, and you can rename or even move the site folder. After renaming or moving the folder, you must open the site in its new location before you can connect to it.

Opening a Site

To open a site, complete the following steps:

1. From the **File** menu, click **Site Selection**.

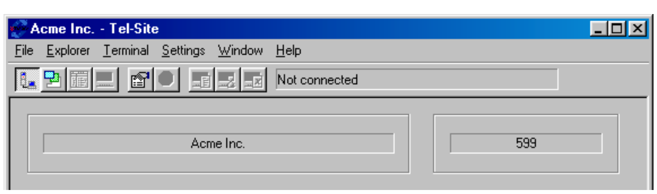
   - The **Site Selection** window appears.

2. Select a site from the **Site Name** column.

3. Click **Open**.

   - The site opens, with its name displayed in the **Connection** window.

Site name displayed in Connection window



CONNECTION METHODS

Tel-Site offers various methods for connecting to a TVS. One method is available for connecting to a site locally (on-site) and three methods are available for connecting to a site remotely (off-site).

- Tel-Site Modem to TVS Modem through two PEU ports (on-site access)

- TVS Modem as an Extension (off-site access)
- Dedicated CO Line to TVS Modem (off-site access)
- TVS Modem Shares a CO Line (off-site access)

Note The TVS is configured by default to support an on-site connection. This allows for the initial connection to the server in order to configure it for one of the three remote methods.

Tel-Site Modem to TVS Modem through two PEU ports (on-site)

The method for the on-site connection is a modem-to-modem connection through two Telecor Port Expansion Unit (PEU) extension ports.

TVS Configuration

If connecting to the TVS for the first time, the TVS Configuration steps do not need to be followed, as this configuration is enabled by default. Refer to these steps only if required to revert back to an on-site connection. The on-site configuration tells the modem when to answer an incoming call and ensures port 16 is defined as a Modem/Fax Port with extension 599.

1. With the site open, through Tel-Site, click the **Configuration** window button in the toolbar.

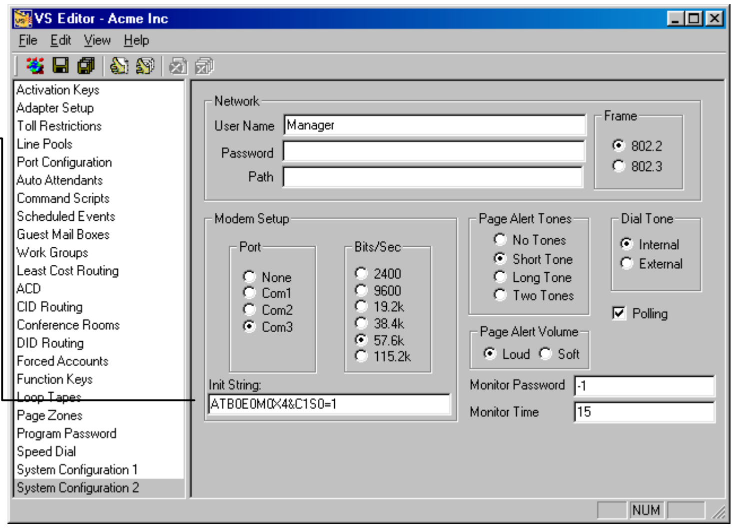
2. Click the **VS1 Editor** button.

    - The VS1 Editor configuration program opens.

3. In the Tree Control display, click **System Configuration 2.**

4. In the **Init String:** text box, check that the end of the modem initialization string reads **S0=1**. (**S0=1** sets the modem to answer on the first ring.) By default, this setting is already in place.

Modem initialization string



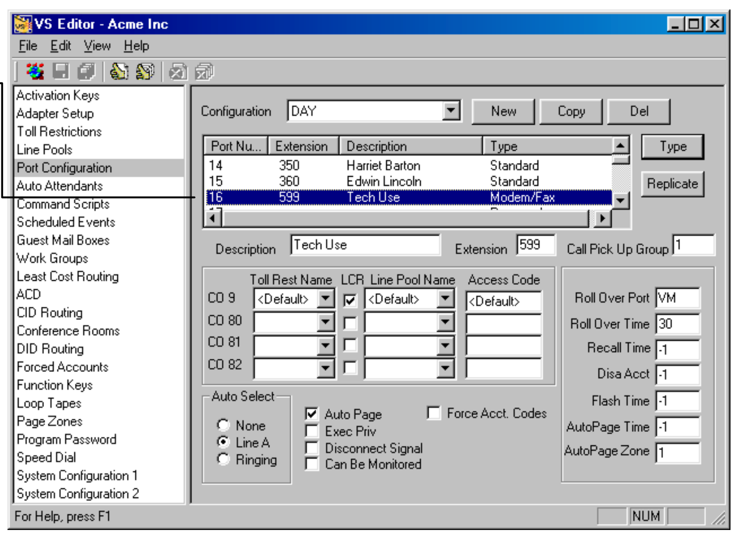
Modem initialization string for Tel-Site Modem to TVS Modem through two PEU Extension Ports connection

5. Click the **Save** button in the toolbar.

6. In the **Tree Control** display, click **Port Configurations**.

7. In each configuration, make sure that the extension port connected to the TVS modem is configured as a Modem/Fax port with the extension 599 (the extension number 599 is reserved for a Modem/Fax port). By default, extension port 16 is configured in this manner.

Modem/Fax port



Modem/Fax Port Configuration

8. Click the **Save** button in the toolbar.

RESET REQUIRED!

9. If steps 4 or 7 had to be changed from the default settings, use your current connection method to upload the new settings to the customer TVS and reload the changes. *See "Tel-Site Basic Operation" on page 76*.

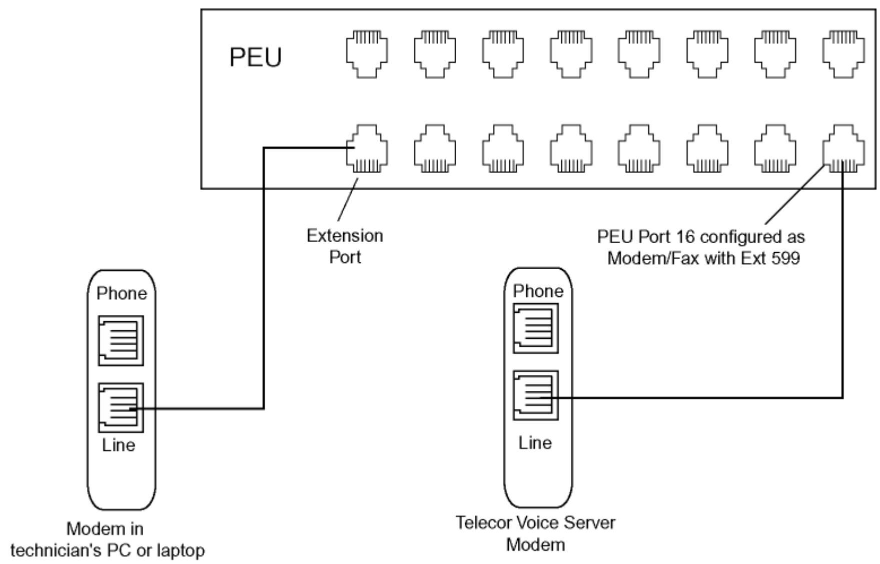
Hardware Setup

Prior to making the necessary hardware changes, ensure the TVS is configured to support the **Tel-Site Modem to TVS Modem through two PEU Ports** connection (see page page 58).

The diagram shows an on-site modem-to-modem connection through two PEU ports. A standard telephone cord from the TVS modem is run to PEU Port 16, which is configured by default as a Modem/Fax port with the extension number 599. Another telephone cord is run from the Tel-Site laptop or PC to an unused extension port on the PEU.

Alternatively, the telephone cord from the Tel-Site computer does not need to be directly connected to the PEU. It can be plugged into the data port of a Telecor DP200 Display Phone, or into a wall jack of a station extension. The only requirement is that the cord is connected to an extension somewhere on the system.

Tel-Site Modem to TVS Modem through Two PEU Extension Ports (on-site access)

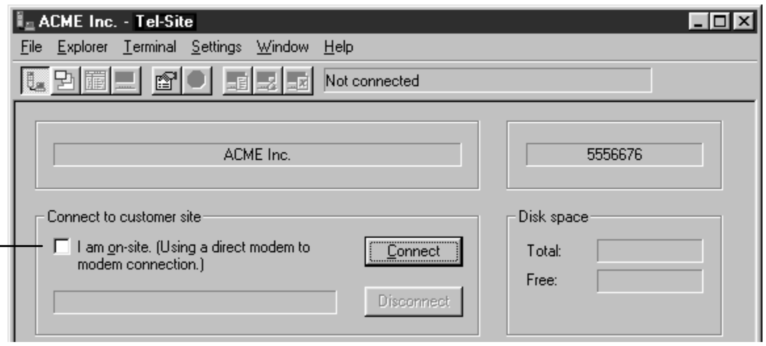


Tel-Site Setup

The connection between Tel-Site and the PEU extension port is made in much the same way as an internal call, only a modem is answering instead of a station user. The Tel-Site application needs to know what extension is going to answer the call.

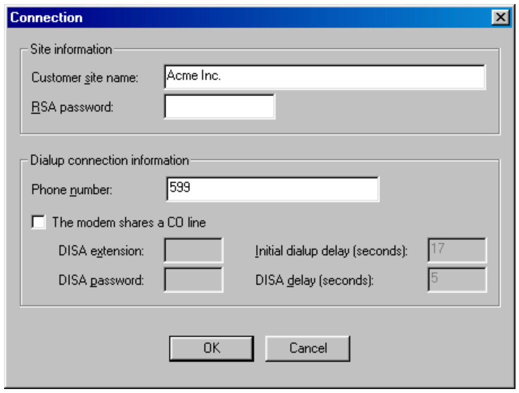
1. In the **Connection** window, ensure the **I am on-site**. **(Using a direct modem to modem connection.)** check box is cleared. Although you are on-site, the connection is made through an extension port on the PEU, not directly to the modem in the TVS.

Ensure this box is cleared



Connection window

2. From the **File** menu, click **Site Selection**.
   - The **Site Selection** window appears.
3. Select the site from the **Site Name** column.
4. Click **Edit Site**.
   - The **Connection** dialog box appears.
5. If an RSA password is required to enable the Tel-Site modem to communicate with the TVS, enter it in the **RSA password** text box. *See "Remote System Access (RSA) Password" on page 78 for more information*. If connecting for the first time, leave this text box blank
6. In the **Phone number** text box, enter 599. Extension 599 is the default extension number of Port 16, which is connected to the TVS modem.
7. Ensure the check box **The modem shares a CO line** is cleared.



Setup in Connection dialog box for Tel-Site Modem to TVS Modem through two PEU Extension Ports connection

8. Click **OK** to return to the **Site Selection** window.

9. Click **OK** to return to the **Connection** window.

10. See *"Tel-Site Basic Operation" on page 76* to:

- Connect to a site
- Download files from a site
- Change site configurations
- Assign an RSA (Remote System Access) password
- Upload files to a site
- Reload configuration changes

Tel-Site Modem to TVS Modem through Two PEU Extension Ports Connection Checklist

❑ In the **System Configuration 2** pane of the VS1 Editor program, set the modem to answer on the first ring.

❑ In each configuration, make sure that PEU Port 16 is configured as a Modem/Fax port with the extension 599.

❑ Using your current connection method, upload the new configuration settings (only if changed from default) to the customer TVS and reload the changes.

❑ Connect a standard telephone cord between the line jack of the TVS modem and PEU port 16.

❑ In the Tel-Site **Connection** window, verify that the **I am on-site (Using a direct modem to modem connection.)** check box is ***not*** selected.

❑ In the **Connection** dialog box of the **Site Selection** window, enter an RSA password if required.

❑ In the **Phone Number** text box, enter 599. Ensure **The modem shares a CO line** check box is cleared.

TVS Modem as an Extension Connection (off-site access)

The setup for this method is similar to the on-site modem to modem connection through two PEU extension ports. The difference is that Tel-Site is calling in from a remote site and transferred (either automatically by an Auto Attendant or manually by a receptionist) to the PEU extension port connected to the TVS modem.

TVS Configuration

For a TVS Modem as an Extension Connection, the configuration tells the modem when to answer an incoming call and ensures port 16 is defined as a Modem/Fax Port with extension 599.

1. With the site open through Tel-Site, click the **Configuration** window button in the toolbar.

2. Click the **VS1 Editor** button.

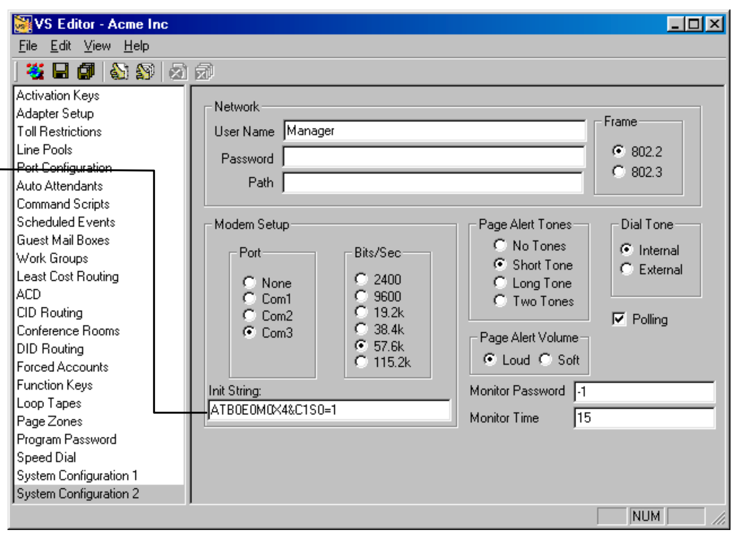
   • The VS1 Editor configuration program opens.

3. In the Tree Control display, click **System Configuration 2.**

4. In the **Init String:** text box, check that the end of the modem initialization string reads **S0=1**. (**S0=1** sets the modem to answer on the first ring.) By default, this setting is already in place.

5. Click the **Save** button in the toolbar.

Modem initialization string

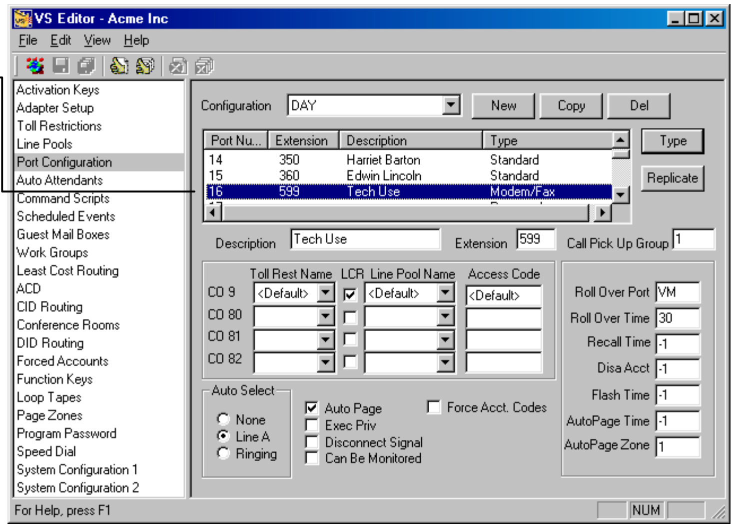


Modem initialization string for TVS Modem as an Extension connection

6. In the **Tree Control** display, click **Port Configurations**.

7. In each configuration, make sure that the extension port connected to the TVS modem is configured as a Modem/Fax port with the extension 599 (the extension number 599 is reserved for a Modem/Fax port). By default, extension port 16 is configured in this manner.

Modem/Fax port



Modem/Fax Port Configuration

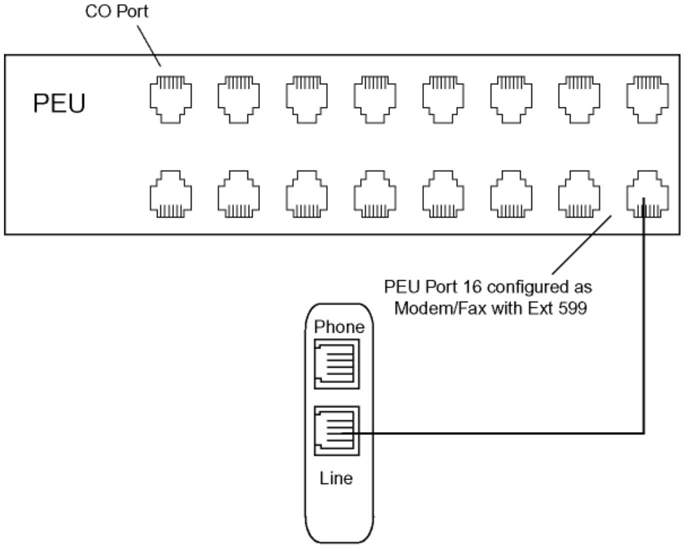
8. Click the **Save** button in the toolbar.

9. If step 4 or step 7 had to be changed from the default settings, use your current connection method (such as the default on-site connection) to upload the new settings to the customer TVS and reload the changes. *See "Tel-Site Basic Operation" on page 76*.

Hardware Setup

Prior to making the necessary hardware changes, ensure the TVS is configured to support the **TVS Modem as an Extension** connection (see page 63).

The diagram below shows the hardware setup for a modem as an extension connection. A standard telephone cord is run from the line jack of the TVS modem to PEU port 16 configured as a Modem/Fax Port with the extension 599.



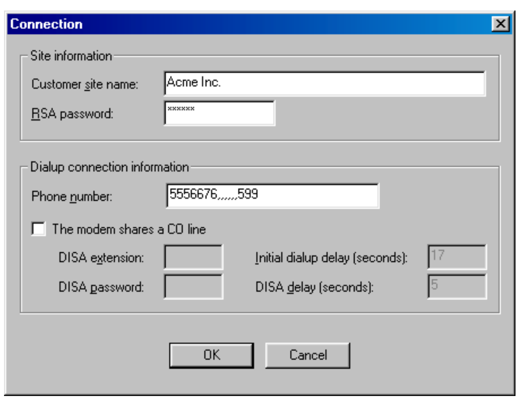
TVS Modem as an Extension (off-site access)

Tel-Site Setup

For a TVS Modem as an Extension Connection, Tel-Site must be set up so that it can dial the customer site phone number. If answered by an Auto Attendant, Tel-Site must automatically dial the extension number 599. If answered by a receptionist, the call must be manually transferred to extension number 599.

1. From the **File** menu, click **Site Selection**.
   - The **Site Selection** window appears.
2. Select the site from the **Site Name** column.
3. Click **Edit Site**.
   - The **Connection** dialog box appears.
4. If an RSA password is required to enable the Tel-Site modem to communicate with the TVS, enter it in the **RSA password** text box. *See "Remote System Access (RSA) Password" on page 78 for more information*.
5. In the **Phone number** text box, enter the customer site phone number to dial. If the Tel-Site call is to be answered by an Auto Attendant, enter five or six commas after the phone number and then the extension number 599. Each comma is used as a 1-second delay in the dialing process to give the Auto Attendant time to answer before it receives the digits 599. The number of commas used may vary depending on how long it takes for the Auto Attendant to answer.

   If the Tel-Site call is to be answered by a receptionist, you must contact the receptionist beforehand and let her know that a modem call will be made. Inform her that when she answers the modem call to please transfer it to extension 599.
6. In the **Dialup connection information** group box, ensure **The modem shares a CO line** check box is cleared.



Connection dialog box for TVS Modem as an Extension connection with Auto Attendant answering

7. Click **OK** to return to the Site Selection window.
8. Click **OK** to return to the Connection window.
9. See *"Tel-Site Basic Operation" on page 76* to:
   - Connect to a site

- Download files from a site
- Change site configurations
- Assign an RSA (Remote System Access) password
- Upload files to a site
- Reload configuration changes

TVS Modem as an Extension Connection Checklist

- ❑ In the **System Configuration 2** pane of the VS1 Editor program, set the modem to answer on the first ring.
- ❑ In each configuration, make sure that PEU Port 16 is configured as a Modem/Fax port with the extension 599.
- ❑ Using your current connection method, upload the new configuration settings (only if changed from default) to the customer TVS and reload the changes.
- ❑ Connect a standard telephone cord between the line jack of the TVS modem and PEU Port 16, which is configured by default as a Modem/Fax port.
- ❑ In the **Connection** dialog box of the **Site Selection** window, enter an RSA password if required.
- ❑ Enter a phone number of the CO line plugged into the line jack of the TVS modem.
- ❑ If an Auto Attendant will answer the Tel-Site call, enter five or six commas after the phone number and then the extension number 599.
- ❑ Ensure **The modem shares a CO line** check box is cleared.

Dedicated CO Line to TVS Modem Connection (off-site access)

With this connection method, the Tel-Site application dials the site and waits for the TVS modem to answer and connect. The advantage of having a dedicated CO line for the TVS modem is the ease of use and reliability of the connection. The disadvantage is that the CO line cannot be used for anything other than the TVS modem.

TVS Configuration

For a Dedicated CO to Modem Connection, you must configure the TVS modem to answer the Tel-Site modem call.

1. With the site open through Tel-Site, click the **Configuration** window button in the toolbar.

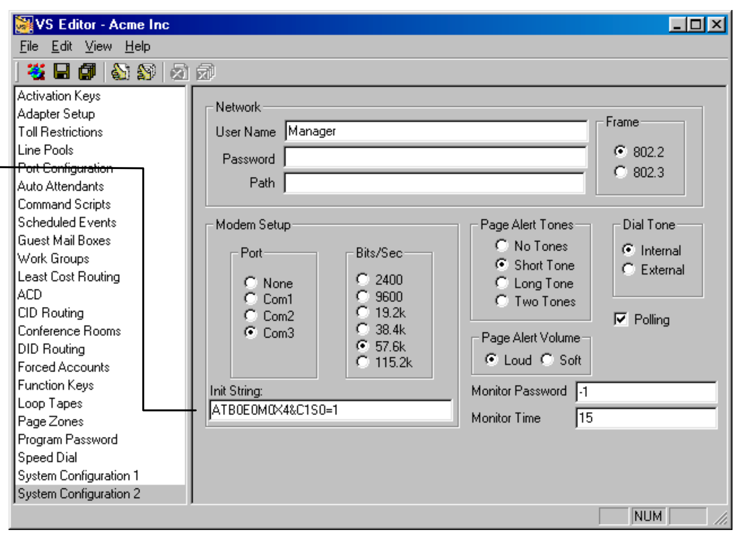
2. Click the **VS1 Editor** button.

   - The VS1 Editor configuration program opens.

3. In the Tree Control display, click **System Configuration 2.**

4. In the **Init String:** text box, check that the end of the modem initialization string reads **S0=1**. (**S0=1** sets the modem to answer on the first ring.) By default, this setting is already in place.

Modem initialization string



Modem initialization string for Dedicated CO Line to TVS Modem connection

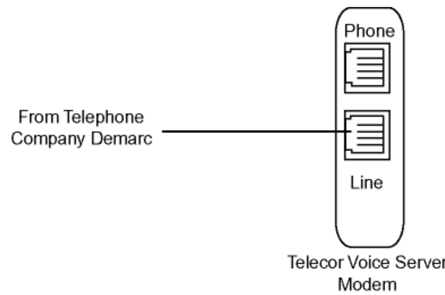
5. Click the **Save** button in the toolbar.

6. If step 4 had to be changed from the default setting, use your current connection method (such as the default on-site connection) to upload the new settings to the customer TVS and reload the changes. *See "Tel-Site Basic Operation" on page 76.*

Hardware Setup

Prior to making the necessary hardware changes, ensure the TVS is configured to support the **Dedicated CO Line to TVS Modem** connection (see page 67).

The diagram below shows the hardware setup for a dedicated CO line connection. A CO line from the telephone company demarc is run to the line jack of the TVS modem.



Dedicated CO Line to TVS Modem (off-site access)

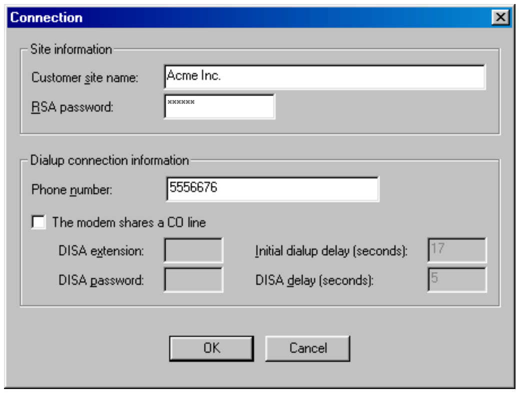
Tel-Site Setup

For a Dedicated CO Line to TVS Modem connection, Tel-Site must be set up so that it can dial the phone number of the CO line plugged into the line jack of the TVS modem.

1. From the **File** menu, click **Site Selection**.
   - The **Site Selection** window appears.
2. Select the site from the **Site Name** column.
3. Click **Edit Site**.
   - The **Connection** dialog box appears.
4. If an RSA password is required to enable the Tel-Site modem to communicate with the TVS, enter it in the **RSA password** text box. *See "Remote System Access (RSA) Password" on page 78 for more information*.
5. In the **Phone number** text box, enter the phone number of the CO line plugged into the line jack of the TVS modem.

Note	When specifying the customer site phone number, you can either specify the number explicitly (using a 9 to get an outside line, for example) or use Standard Number Format. See the Tel-Site Help topic **Standard Number Format** for more information.

6. In the **Dialup connection information** group box, ensure **The modem shares a CO line** check box is cleared.



Setup in Connection dialog box for Dedicated CO Line to TVS Modem connection

7. Click **OK** to return to the Site Selection window.

8. Click **OK** to return to the Connection window.

9. See *“Tel-Site Basic Operation” on page 76* to:

- Connect to a site
- Download files from a site
- Change site configurations
- Assign an RSA (Remote System Access) password
- Upload files to a site
- Reload configuration changes

Dedicated CO Line to TVS Modem Connection Checklist

❑ In the **System Configuration 2** pane of the VS1 Editor program, set the modem to answer on the first ring.

❑ Using your current connection method, upload the new configuration setting (only if changed from default) to the customer TVS and reload the changes.

❑ Connect the CO line from the telephone company demarc to the line jack of the TVS modem.

❑ In the **Connection** dialog box of the **Site Selection** window, enter an RSA password if required.

❑ Enter the phone number of the CO line plugged into the line jack of the TVS modem.

❑ Ensure **The modem shares a CO line** check box is cleared.

TVS Modem Shares a CO Line Connection (off-site access)

Sharing a CO line for a connection is a common way to access remote sites. This section describes the TVS configuration required for a successful connection. In addition, you learn why Tel-Site uses DISA (Direct Inward System Access) accounts and where to enter that information.

Note The modems used for shared CO connections must support partial dialing. Partial dialing enables Tel-Site to dial the customer site, delay long enough for the TVS to answer, and then send a Direct Inward System Access (DISA) logon password to complete the connection. *For more information, see the **Modem Properly Installed** topic in the Tel-Site **Help** menu.*

About Direct Inward System Access (DISA)

The **TVS Modem Shares a CO Line** connection uses DISA (Direct Inward System Access) as part of its logon to the system. DISA is a way to gain access to the internal dial tone of the VS1 telephone system from an external phone not connected to the system. For example, you are off-site and want to access an outside line from the office. You perform the normal DISA steps of pressing #, then your extension, and then your DISA account.

DISA for the Tel-Site application uses the same procedures as normal DISA activity. The only difference is that Tel-Site performs the steps automatically to connect.

TVS Configuration

For **TVS Modem Shares a CO Line** connection, the following steps must be completed:

- Configure the TVS modem to not answer the Tel-Site modem call.
- Enable DISA on the CO port connected to the TVS modem.
- Set up a DISA account on an extension port type

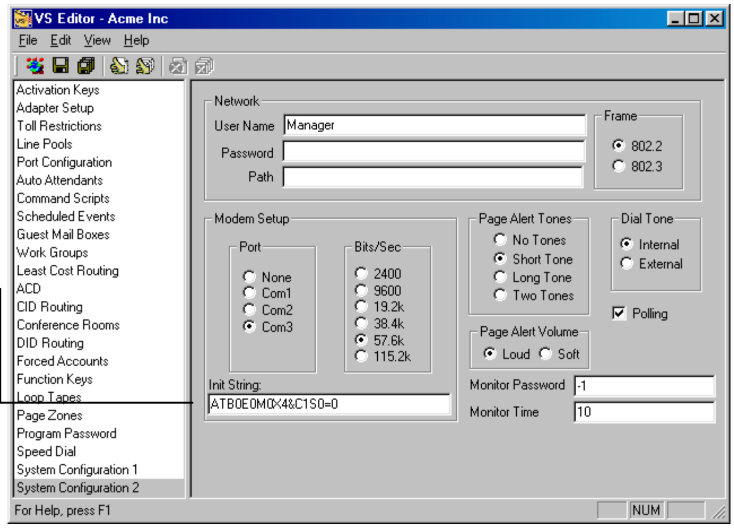
With the site open through Tel-Site, click the **VS1 Editor** button in the **Configuration** window. This opens the VS1 Editor configuration program to make the above changes.

Configure the TVS modem to not answer the Tel-Site modem call

1. In the Tree Control display, click **System Configuration 2**.

2. In the **Init String:** text box, replace the current **S0=[n]** with **S0=0** at the end of the modem initialization string. (Typing S0=0 sets the modem so that it does not answer.)

3. Click the **Save** button in the toolbar.

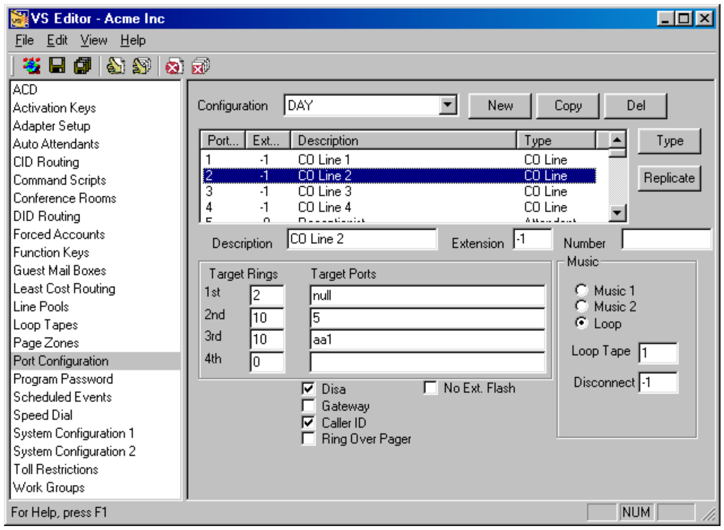


Modem initialization string

Modem initialization string for TVS Modem Shares a CO Line connection

Enabling DISA on the CO Port connected to the TVS modem

1. In the VS1 Editor Tree Control display, click **Port Configurations**.
2. In the **Port Configurations** pane, select the configuration where you want to make changes. For example, select the DAY configuration.
3. Select the CO port that is shared with the TVS modem.
4. Under **Rings**, set the CO port to ring the first target for two rings.
5. Under **Target Ports**, set the target by typing **null**. (The **null** device doesn’t ring a physical extension. It gives Tel-Site time to send the DISA sequence to the TVS.)
6. Set the number of rings for the remaining targets according to your configuration.
7. Select the **DISA** check box.
8. Make sure you replicate the changes made to this port in all configurations, not just for the DAY configuration.



DISA setup on CO Port

Setting up a DISA Password on an Extension Port

1. In the VS1 Editor Tree Control display, click **Port Configurations**.

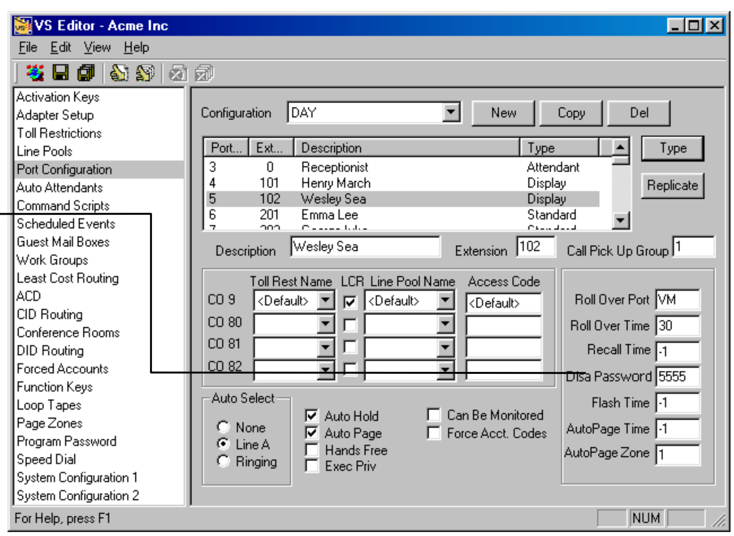
2. In the **Port Configurations** pane, select the configuration where you want to make changes. For example, select the DAY configuration.

3. In the **Port Configurations** pane, select the Extension port of a user you want to give a DISA password.

4. In the **DISA Password** text box, enter a five-digit number ranging from 00000–99999. By default, the DISA Password is set to –1, which means no password has been assigned, and no DISA features are available for that extension.

5. Make sure you replicate the changes made to this port in all configurations, not just for the DAY configuration.

DISA Password text box



DISA Password on extension port

Uploading new settings and reloading changes

RESET REQUIRED!

After the above settings have been configured, use your current connection method (such as the default on-site connection) to upload the new settings to the customer TVS and reload the changes. *See "Tel-Site Basic Operation" on page 76.*

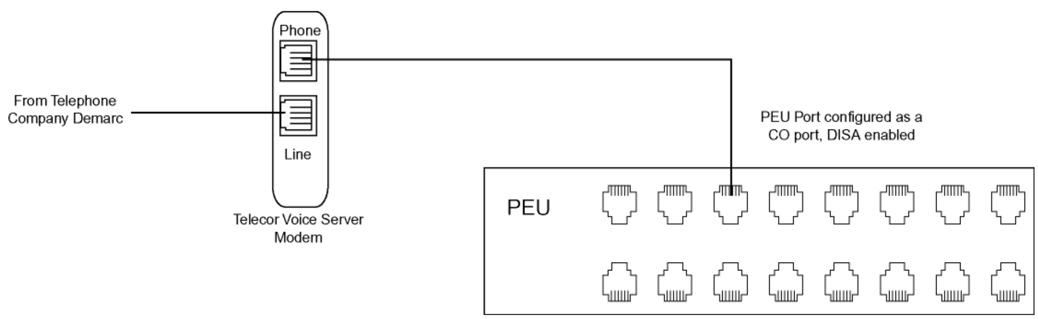
Hardware Setup

Prior to making the necessary hardware changes, ensure the TVS is configured to support the **TVS Modem Shares a CO Line** connection (see page 70).

The diagram below shows the hardware setup for a shared CO connection. The CO line from the telephone company demarc is run to the line jack of the TVS modem. Another telephone cord is run from the phone jack of the TVS modem to a CO port on the PEU.

Note:	It is important that you share one of the least-used CO lines. Usually a CO line in the middle of the inbound hunt group and outbound line pool is best.

TVS Modem Shares a CO Line (off-site access)



Tel-Site Setup

Tel-Site must be set up so that it can dial the phone number of the CO Line plugged into the line jack of the TVS modem. It then must automatically dial the extension number set up with DISA and the associated DISA account to complete the connection. Complete the following steps:

1. From the **File** menu, click **Site Selection**.
   - The **Site Selection** window appears.
2. Select the site from the **Site Name** column.
3. Click **Edit Site**.
   - The **Connection** dialog box appears.
4. If an RSA password is required to enable the Tel-Site modem to communicate with the TVS, enter it in the **RSA password** text box. *See "Remote System Access (RSA) Password" on page 78 for more information*.

5. In the **Phone number** text box, enter the phone number of the CO line plugged into the line jack of the TVS modem.

Note	When specifying the customer site phone number, you can either specify the number explicitly (using a 9 to get an outside line, for example) or use Standard Number Format. See the Tel-Site Help topic **Standard Number Format** for more information.

6. In the **Dialup connection information** group box, select **The modem shares a CO line** check box.

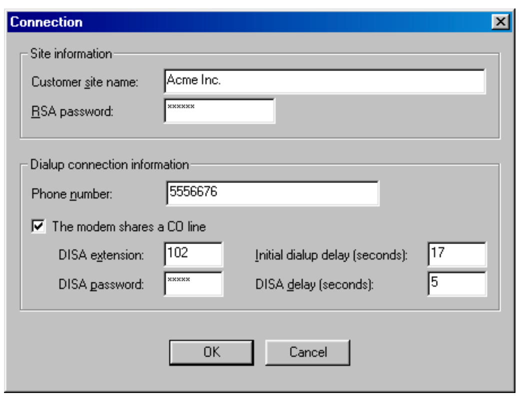
7. In the **DISA extension** text box, enter an extension set up with a DISA password.

8. In the **DISA password** text box, enter the associated DISA password for the extension.

Note 1	Generally, the default value of five seconds for the **DISA delay** does not need to be changed. However, the **Initial dialup delay** may need to be changed. If the default setting does not work, connect a phone to the phone jack on the modem and listen in parallel while the modem tries to connect. Count the number of seconds between the time of the last digit dialed and after the first ring back, and then enter that number in the **Initial dialup delay** text box.

Note 2	If your modem speaker is set loud enough for you to hear the ring backs, you can listen for the Voice Server ring back, and then press **Shift** while clicking the **Connect** button. This will cause Tel-Site to send the DISA logon immediately instead of waiting for the **Initial dialup delay** time to expire.

Setup in Connection dialog box for TVS Modem Shares a CO Line connection



9. Click **OK** to return to the **Site Selection** window.

10. Click **OK** to return to the **Connection** window.

11. See *"Tel-Site Basic Operation"* on page 76 to:

- Connect to a site
- Download files from a site
- Change site configurations
- Assign an RSA (Remote System Access) password
- Upload files to a site
- Reload configuration changes

TVS Modem Shares a CO Line Connection Checklist

- ❑ In the **System Configuration 2** pane of the VS1 Editor program, set the modem so it does not answer.
- ❑ In each configuration, enable DISA on the CO Port connected to the TVS modem.
- ❑ On the same extension port in each configuration, set up a DISA password.
- ❑ Using your current connection method, upload the new configuration to the customer TVS and reload the changes.
- ❑ Connect the CO line from the telephone company demarc to the line jack of the TVS modem.
- ❑ Connect a standard telephone cord from the phone jack of the TVS modem to a DISA-enabled CO port on the PEU.
- ❑ In the **Connection** dialog box of the **Site Selection** window, enter an RSA password if required.
- ❑ Enter the phone number of the DISA-enabled CO line plugged into the line jack of the TVS modem.
- ❑ Select **The modem shares a CO line** check box.
- ❑ Enter the extension set up with a DISA password. Enter the associated DISA password for the extension.

TEL-SITE BASIC OPERATION

The Tel-Site system management application enables you to make site changes in five steps after you have set up a profile for that site. These steps are:

1. Connecting to a site.
2. Downloading any files that have changes from a site.
3. Changing those files on your local computer using VS1 Editor.
4. Uploading those changes to the site.
5. Reloading those changes to put them in effect

Connecting to a Site

The **Connection** window for Tel-Site appears when the application is first started. Before you can connect to a site using Tel-Site, you must first make hardware, wiring, and system configuration changes. *See "Connection Methods," page 58.*

1. From the **File** menu, click **Site Selection**.

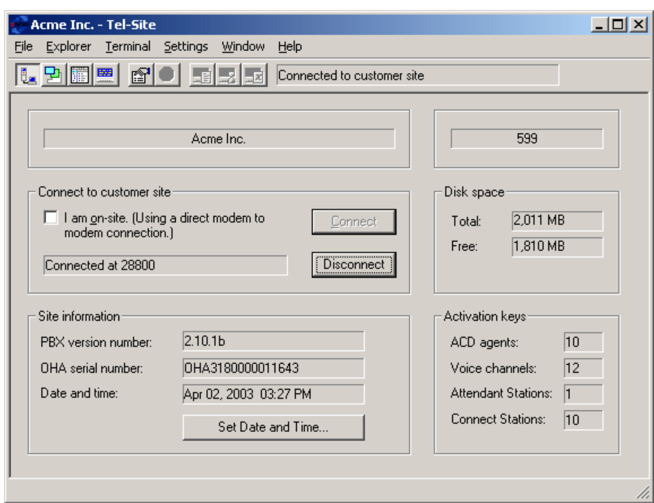
   - The **Site Selection** window appears.

2. Select the site from the **Site Name** column.

3. Click **Open**.

4. In the **Connection** window, click **Connect**.

5. Tel-Site dials the site, logs on, and supplies any necessary passwords to complete the connection.



Connection window

Downloading Files from a Site

When connected to a customer site and the **Configuration** window is opened, Tel-Site scans and displays any TVS files that have changed since the last remote activity. It can then download any updated files so that your local computer matches the latest customer site configuration.

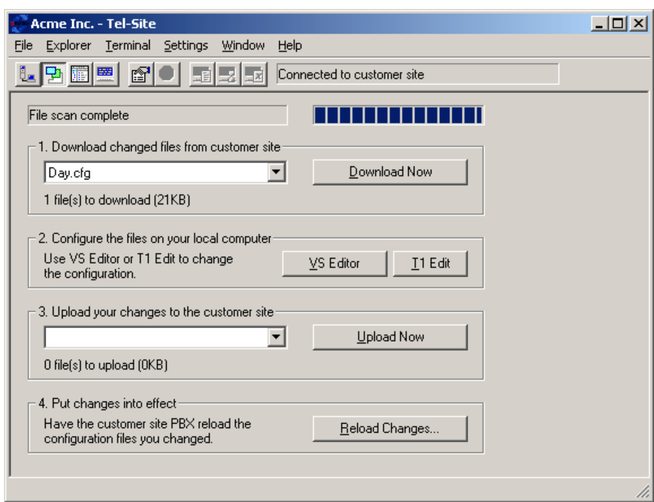
Note The first time you open the **Configuration** window after upgrading the TVS software, a dialog box opens indicating an upgrade has occurred. You must download all of the Configuration files so they correspond with the latest VS1 software version.

1. On the toolbar, click the **Configuration** window button.

2. Wait for Tel-Site to scan the files on the TVS. After scanning, files that need to be downloaded appear in the **Download changed files from customer site** drop-down list box. If no files have changed, the drop-down list box is empty.

3. Click **Download Now** to download the files to your local computer. For example, the screen below shows file **Day.cfg** has changed since the last remote activity.

Warning! Clicking the **Download Now** button when there are no files present causes Tel-Site to download all the configuration files from the site. This should be done only if you believe the Tel-Site application failed to detect changed files.



Download remote files

Changing Site Configurations

Configuration changes are performed on copies of site files on the Tel-Site computer. After any changed customer site files are downloaded to the Tel-Site computer to match the latest customer site configuration, the **VS1 Editor** program is used to configure the files.

1. In the **Configuration** window, click the **VS1 Editor** to open the VS1 Editor configuration program.

2. Change the system configuration files as needed. *See the VS1 Editor section for information .*

Remote System Access (RSA) Password

The TVS can be set up with a password to enable a remote modem to communicate with the system. To assign a RSA password to the TVS, complete the following steps.

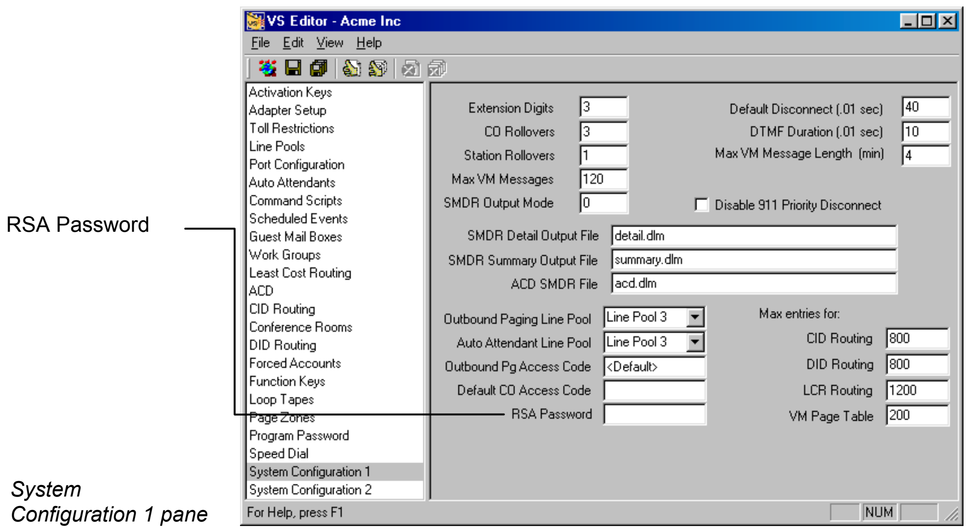
1. In the Tree Control display of the VS1 Editor program, click **System Configuration 1.**

2. In the **RSA Password** text box, assign a password. The password can be alphanumeric. Leaving the box blank means there is no password.

3. Click the **Save** button in the toolbar.

RESET REQUIRED!

4. Upload the password to the customer TVS and reload the changes (see page 79).

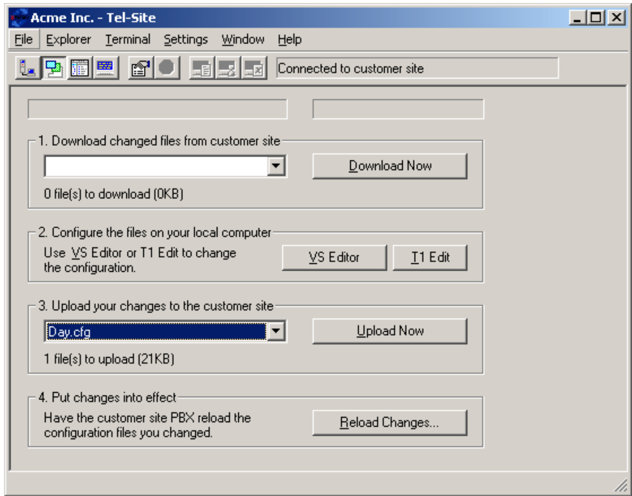


Uploading Files to a Site

After making changes using **VS1 Editor**, the drop-down list box for **Upload Now** in the **Configuration** window shows the configuration files that have changed. For example, **Day.cfg**.

1. Click **Upload Now** to upload the files to the TVS at the customer site.

Warning! Clicking **Upload Now** when there are no files listed in the **Upload your changes to customer site** drop-down list box causes Tel-Site to upload all the configuration files to the site. You should do this only if you think Tel-Site failed to detect one or more changed files on your computer.



Upload files

Reloading Configuration Changes

Some configuration changes require a reset of the TVS. Resets should be made during periods of low call activity, such as after-business hours. Call traffic is interrupted when the TVS is reset. However, not all configuration changes require a system reset. To instruct the customer TVS to put the changes in effect, complete the following steps:

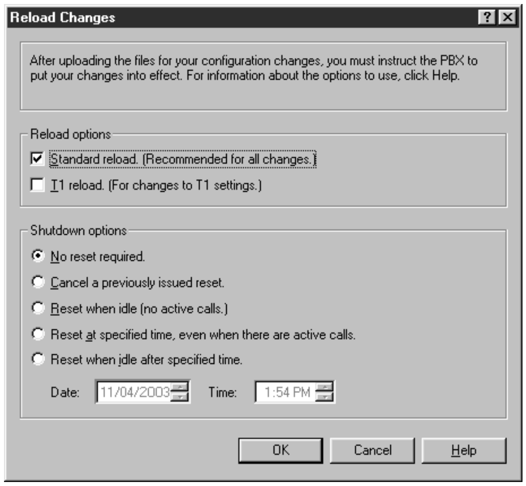
1. Click **Reload Changes**.

    - The **Reload Changes** dialog box appears.

2. In the **Reload options** group box, you can choose between a **Standard reload**, which is recommended for all changes, or a **T1 reload**, which is used for changes to T1 settings.

3. The **Shutdown options** group box is where the TVS is instructed to either reset to put configuration changes into effect, or not to reset because the configuration changes made do not require a system reset.

    If a reset of the TVS is required, the reset should be made during periods of low call activity, such as after business hours. Call traffic is interrupted when the TVS is reset.



Reload changes

System Reset Required

If any of the following items are changed with the VS1 Editor and uploaded to the server, a system resest is required.

- ACDs
- Adapter Setup
- Activation Keys
- Auto Attendants (except extension numbers)
- Auto-Select in Extension Port Configuration
- AutoPage in Extension Port Configuration
- Conference Rooms
- Function Keys
- Forced Account Codes[1]
- Guest Mailboxes
- Handsfree in Extension Port Configuration
- Loop Tapes
- Page Zones
- Port Types
- System Configuration 1-2
- Work Groups

[1] A System Reset for Forced Account Codes is only required if changing the number of digits or changing between verified and non-verified.

VS1 Editor

OVERVIEW OF VS1 EDITOR

The VS1 Editor application allows a VS1 Telephone System to be configured for operational use. All system configuration settings and properties are set up and controlled using this application. VS1 Editor can be opened independently or can be opened through Tel-Site via the Configuration window.

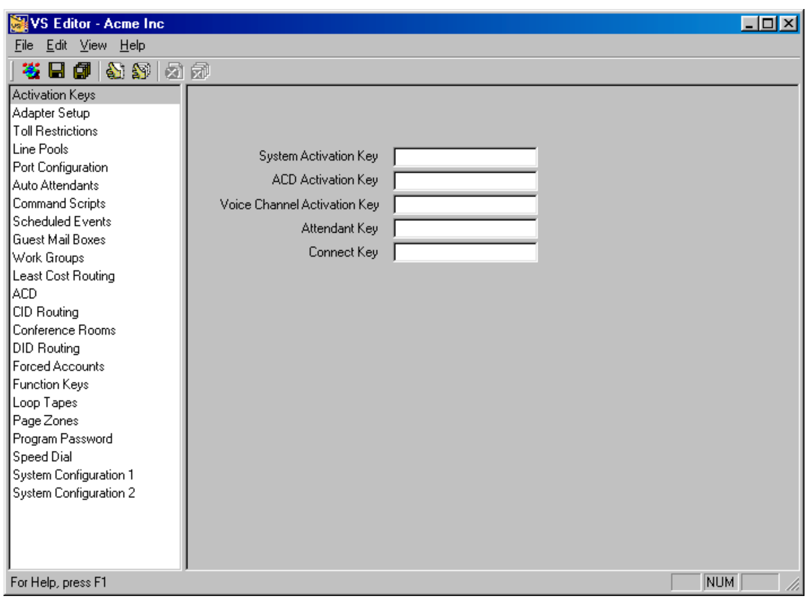
The VS1 Editor application is available for download from the VS1 Dealer Page of the Telecor Web Site (www.telecor.com). Please visit the page for details on licensing information and installing the application.

In addition, the VS1 Editor can be installed with limited features to allow end-users the ability to make common changes to the VS1 system without risk of modifying or damaging important configuration data *(see VS1 Editor – Limited Feature Version, page 168)*.

The VS1 Editor application is a Windows®-based program that operates on Microsoft® Windows® 98, Windows® 2000, and Windows® XP operating systems.

The left pane consists of the Tree Control, which by default displays items in a logical order. Logical order means that the items are displayed in an ideal top-to-bottom sequence that can be followed when configuring a VS1 System. Clicking on an item brings up its accompanying pane to the right.

The order in which the items are displayed in the Tree Control can be changed from logical to alphabetical by selecting **View > Order** from the menu bar. The order in which each pane is described in this section is Alphabetical



Activation Key pane appears when VS1 Editor is opened

ACTIVATION KEYS

RESET REQUIRED!

The VS1 system controls software features through the use of Activation Keys. Activation Keys “turn on” or activate certain VS1 phone system options purchased from Telecor. Activation Keys are linked to the Telecor Voice Server (TVS) System ID Number (OHA Serial Number) and only activate software installed on that TVS.

The software options that require Activation Keys include:

- System Activation Key (version specific)
- Automatic Call Distribution (ACD) Agent Package (for every 5 agents)
- Voice Channel Activation (for every 4 voice channels)
- Telecor Attendant CTI client application
- Telecor Connect CTI client application

Receiving Activation Keys

Each software option has its own Activation Key Request card that arrives in the box shipped to you from your distributor. There are four steps you must follow in order to receive your software Activation Keys from Telecor.

1. Review the Activation Key Request card number for each option purchased. The Activation Key Request card number is a 13-digit number similar in pattern to the following example: 000-00000-00000.

2. Call Telecor Technical Support.

3. Give the Technical Support representative the Activation Key Request card number and the System ID Number (OHA Serial Number) for the TVS on-site.

   - OHA numbers can be found in three areas: the shipping package; in the **Connection** window of Tel-Site, or by typing **sysinfo** at the TVS Command prompt.

4. The Technical Support representative generates a 12-digit Activation Key for each option.

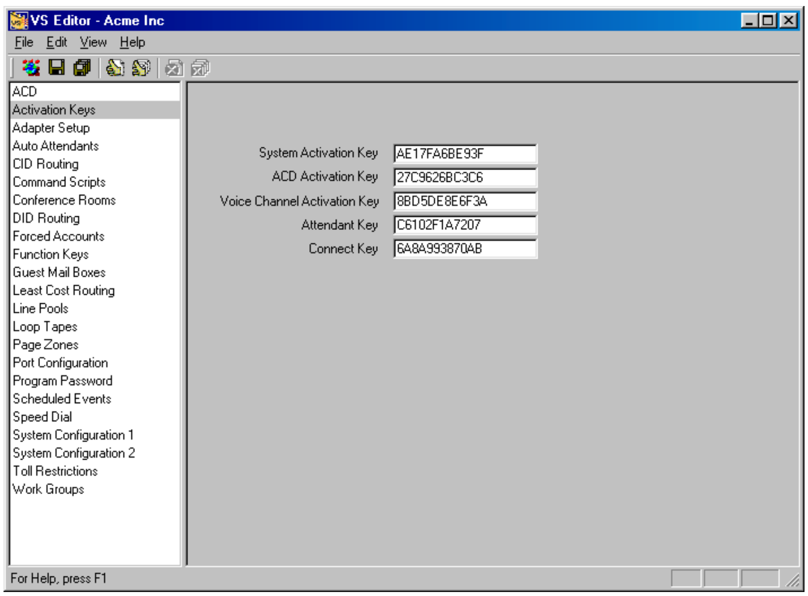
Installing Activation Keys

1. Click **Activation Keys** in the Tree Control display.

2. In the **Activation Keys** pane, type the 12-digit code in the appropriate software option text boxes.

3. Click the **Save** button in the toolbar.

Note	Each Host Adapter Card has a unique Serial Number. The TVS uses the Serial Number of the first Host Adapter Card (ports 1-32) as the System ID Number. The System ID Number is used to authenticate Activation Keys for that particular TVS. If the Host Adapter Card is configured for ports 1-32 is replaced, you must call Technical Support for new Activation Keys.



Activation Key pane

ADAPTER SETUP

RESET REQUIRED!

Adapter Setup is required to indicate to the VS1 System what hardware is allocated for its ports. If upgrading system hardware, Adapter Setup can save you time by enabling you to perform the following actions before installing the new hardware:

- Change port types
- Change configurations
- Upload those configurations using Tel-Site

Complete the following steps:

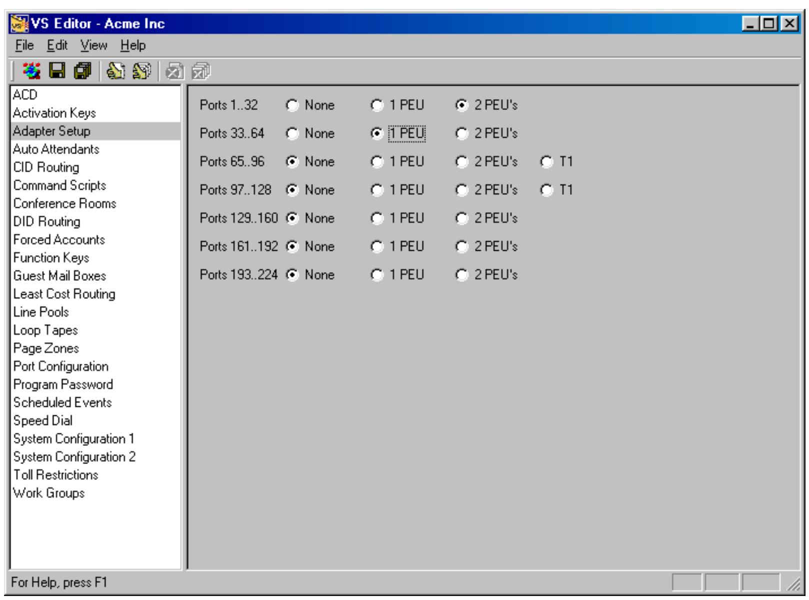
1. Select **Adapter Settings** in the Tree Control display.

    - The **Adapter Settings** pane appears.

2. For Ports **1..32** select if one or two PEUs are used. Repeat this step for additional ports.

3. If one T1 Card is installed, select Ports **97..128**. If a second T1 Card is installed, select Ports **65..96**.

4. Click the **Save** button in the toolbar.



Adapter Setup pane

Note	Ports 193 to 224 are not applicable with 2.10 software and are reserved for future use.

AUTOMATIC CALL DISTRIBUTION

RESET REQUIRED!

Automatic Call Distribution distributes incoming calls in a logical pattern to available company personnel. Incoming CO lines can be routed directly to an ACD or calls can also be transferred manually to an ACD extension from any VS1 station or Auto Attendant. The ACD feature immediately routes the calls to the first available agent. If the agent cannot immediately pick up the call, the ACD system plays a message defined as the Primary Loop Tape. For example, the message could say: “Thank you for calling XYZ, Inc. All our agents are helping other callers. Please hold for the next available agent.”

Each caller hears the Primary Loop Tape message from the beginning. If the Primary Loop Tape is currently being played, a new caller continues to hear ringing until the loop starts over. When the Primary Loop Tape message concludes, the ACD feature automatically plays the Secondary Loop Tape, if it is available. The Secondary Loop Tape message may consist of music or promotional information. If a Secondary Loop Tape message is not available, the system can play music on-hold from the music source connected to the dry contacts on the Port Expansion Unit Model 205 (PEU-205) or the Dry Contact Unit (DCU). If no music is available, the caller hears silence. Both Loop Tape messages are interrupted immediately when an agent answers the call.

There are 10 ACDs available on the VS1 System. A maximum of 95 agents can be logged on to the 10 ACDs, and a maximum of 30 callers can be in the ACD queue at any given time. The definition of a logged on agent is a station option that is logged on to one or more ACDs. For example, if an agent is logged on to three ACDs from a DP200 display phone, that agent is considered only as one agent logged on to the system.

Call Distribution: Within an ACD, the standard method of call distribution is to route incoming calls to the highest priority agent available (top-down distribution). If all available agents have the same priority level, the incoming calls are routed to each agent in order (round-robin distribution).

Agents can log on to the ACD from any VS1 System station option except the Attendant CTI client application. The system automatically recognizes the logged on station option and routes calls to that station.

With the standard method, a call in the ACD is answered by the highest priority agent available, or if all available agents have the same priority level, to each agent in order.

Warnings: The ACD also generates alert messages based on call criteria such as calls per agent ratio, age of the oldest call in the queue, or if there are callers in queue but no agents logged on. The alerts can be configured in several ways, but one approach is to have a voice message played over the paging system (or over speakerphones) to alert the manager or supervisor.

An Activation Key code is required for ACD use on the VS1 phone system. An ACD Activation Key will activate 5 agents. You will find an Activation Key Request card located in the shipment of equipment from your distributor.

Creating an ACD

1. Click **ACD** in the Tree Control display.

    - The **ACD** pane appears.

2. In the **Select ACD** box, select an ACD you want to configure.

3. In the **Priority Mode** group box, select how the calls are distributed amongst multiple ACDs:

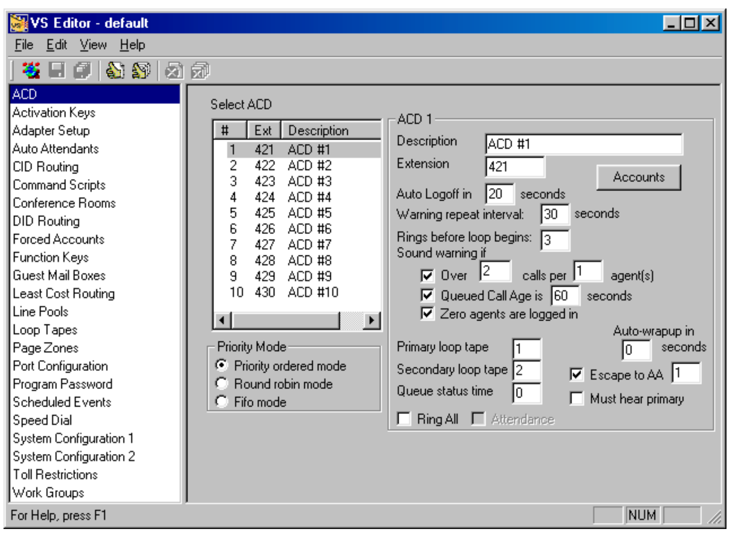
    - **Priority Ordered Mode**: This is the default setting. All calls in ACD #1 are answered, then all calls in ACD #2 are answered, and so on.

    - **Round Robin Mode**: The oldest call in the first called ACD is answered, then the oldest call in the second called ACD is answered, and so on.

    - **FIFO Mode (First In First Out)**: The oldest waiting call is answered first, regardless of the ACD queue that it is in.

Note The difference between Round Robin Mode and FIFO Mode is that in Round Robin Mode, it is possible for the oldest call in the queue not to be answered first depending on what ACD it is in. Whereas, in FIFO Mode, the oldest call is answered first regardless of the ACD it is in.

4. Set up Account Codes for the ACD.



ACD pane

ACD Account Codes

Account Codes are used by agents to log on and log off the ACD system. Each Account Code has an assigned priority to determine how calls in the ACD queue are routed. Up to 100 account codes can be created for each ACD.

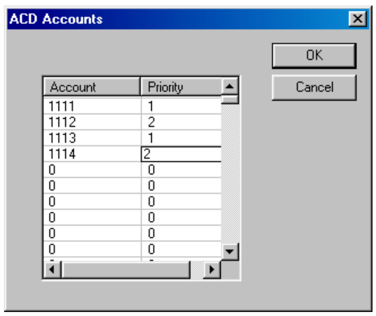
1. To add Account Codes for an ACD, select the ACD in the **Select ACD** box and then click the **Accounts** button.

- The **ACD Accounts** window appears.

2. In a space under the **Account** column enter a unique Account Code of four digits. Do not include spaces or other punctuation.

3. Beside to the newly-created Account code, under the **Priority** column, enter a Priority number of one or two digits. When more than one agent is available on the ACD system, callers are routed to the highest priority agent (top-down distribution). For example, if Agent A has a priority level of 1, and Agent B has a priority level of 2, calls are routed to Agent A first. If all available agents have the same priority level, the incoming calls are routed to each agent in order (round-robin distribution).

ACD Accounts window



4. Click **OK** to return to the **ACD** pane.

Agent Account Codes can be assigned to more than one ACD, enabling that agent to receive calls from all ACDs in which their account code appears. Note that the account codes in multiple ACDs can have different priority levels for each ACD.

ACD No. 1	**ACD No. 2**	**ACD No. 3**	**ACD No. 4**
1111-1	1111-2	1111-3	1111-3
1112-2	1112-1	1112-3	1112-4
1113-4	1113-2	1113-1	1113-3

Configuring an ACD

To configure an ACD enter the following information in the corresponding text boxes. Click the **Save** button in the toolbar when the information has been entered.

Description: Enter a description of the ACD system. This description appears on Attendant and Connect CTI station options and DP200 display phones.

Extension: Enter an extension for the ACD, which is required to transfer callers to the ACD system. The extension must be a valid extension number not already assigned to another station.

Auto log-off in _____seconds: Determines the length of time a station rings before being automatically logged off the ACD. After being logged off, waiting callers are automatically returned to the queue for the next available agent. The caller also maintains their priority status in the queue.

Warning repeat interval _____seconds: Set the repeat interval for ACD Warning Command Scripts.

Rings Before Loop Begins_____: Specify the number of rings the caller hears before being connected to the Primary Loop Tape. If the number of rings is set to 0, the caller continues to hear ringing while in the queue.

Sound Warning If: Determines which call conditions result in a warning announcement being played over a Page Zone. Checking a box will play the warning announcement if the conditions are met. By default, only ACD #1 is set up to announce the warnings. In addition, a Page Zone must be defined for the warning announcement to play over. See the page reference beside each warning for instructions on implementing the warning in other ACDs and defining a Page Zone for the announcement.

Over _____calls per _____agents: *"There are too many calls for the number of agents. Please add another agent."* (page 91)

Queued call age is _____seconds: "*Calls in queue have been waiting too long. Please add another agent."* (page 91)

Zero agents are logged in: *"Calls are in queue and zero agents are logged in. Please add agents."* (page 92)

Primary Loop Tape: Enter the number of the Loop Tape you want callers to hear first. If you enter 0, callers hear only ringing. The Primary Loop Tape should be brief—preferably no longer than 10 seconds. This is because incoming callers hear ringing until the Primary Loop Tape starts over. An example of a short message can be "Thank you for calling XYZ company. All of our agents are busy. Please hold for the next available agent."

Secondary Loop Tape: Enter the number of the Loop Tape for callers to hear after the Primary Loop Tape message concludes. The Secondary Loop Tape message may consist of music or promotional information. Enter **0** for music source 1.

Auto-Wrapup: Enter the time in seconds allowed for the agent to wrap-up the previous call before receiving another call.

Escape to AA: Enter the number of the Auto Attendant (AA1—AA20) that callers access when they press 0 to exit the queue. If this option is not checked, the Escape to AA feature is disabled.

Must Hear Primary: Check this option to force every caller to hear the Primary Loop Tape in its entirety.

Queue Status Time: Enter the time in seconds for how often the status of that ACD queue is written to the ACD Station Message Detail Recorder (SMDR) output file. For example, if you enter 15 in the **Queue Status Time** text box, the status of that queue including number of calls, number of agents in queue, number of agents logged on and so on, is written to the acd.dlm output file every 15 seconds.

Over _____calls per _____agents

If the specified number of calls exceeds the specified number of agents, the following voice file is played: *"There are too many calls for the number of agents. Please add another agent."*

By default, this warning sounds only for ACD #1. In addition, a Page Zone needs to be defined for the warning to sound over. *See page 140 for information on Page Zones.*

This voice file is played as a result of the following default Command Script: ACD1W1.CMD. This Command Script consists of the following system command: **call pager 11** (plays voice file 11, which is in the above quotes). To add a Page Zone to the system command, select the ACD1W1.CMD Command Script in the Command Scripts pane and enter the Page Zone number after **call pager 11**. For example, to have the warning announce over Page Zone 4, the system command in ACD1W1.CMD would read **call pager 11 4**. *See "Command Scripts" on page 116 for more information.*

If the warning is to be implemented in another ACD, then a new Command Script needs to be created recognizing the ACD. For example, if ACD #5 is to announce the warning, then a Command Script titled ACD5W1.CMD needs to be created with the following system command **call pager 11 4** (if 4 is the Page Zone to be used).

Queued call age is _____seconds

If queued call is longer than the seconds specified, the following voice file is played: *"Calls in queue have been waiting too long. Please add another agent."*

By default, this warning sounds only for ACD #1. In addition, a Page Zone needs to be defined for the warning to sound over. *See page 140 for information on Page Zones.*

This voice file is played as a result of the following default Command Script. This Command Script consists of the following system command: **call pager 12** (plays voice file 12, which is in the above quotes). To add a Page Zone to the system command, select the ACD1W2.CMD Command Script in the Command Scripts pane and enter the Page Zone number after **call pager 12**. For example, to have the warning announce over Page Zone 4, the system command in ACD1W2.CMD would read **call pager 12 4**. *See "Command Scripts" on page 116 for more information.*

If the warning is to be implemented in another ACD, then a new Command Script needs to be created recognizing the ACD. For example, if ACD #5 is to announce the warning, then a Command Script titled ACD5W2.CMD needs to be created with the following system command **call pager 12 4** (if 4 is the Page Zone to be used).

Zero agents are logged in

If calls are in the queue and agents are not logged in, the following voice file is played: *"Calls are in queue and zero agents are logged in. Please add agents."*

By default, this warning sounds only for ACD #1. In addition, a Page Zone needs to be defined for the warning to sound over. *See page 140 for information on Page Zones.*

This voice file is played as a result of the following default Command Script. This Command Script consists of the following system command: **call pager 13** (plays voice file 13, which is in the above quotes). To add a Page Zone to the system command, select the ACD1W3.CMD Command Script in the Command Scripts pane and enter the Page Zone number after **call pager 13**. For example, to have the warning announce over Page Zone 4, the system command in ACD1W3.CMD would read **call pager 13 4**. *See "Command Scripts" on page 116 for more information*

If the warning is to be implemented in another ACD, then a new Command Script needs to be created recognizing the ACD. For example, if ACD #5 is to announce the warning, then a Command Script titled ACD5W3.CMD needs to be created with the following system command **call pager 13 4** (if 4 is the Page Zone to be used).

AUTO ATTENDANTS (AA)

RESET REQUIRED! For Extension Numbers only

Auto Attendants are prerecorded voice files that can take the place of human receptionists. Auto Attendants answer and then process incoming calls according to caller DTMF inputs. They provide callers with information in the form of recorded messages and enable callers to route themselves to specific groups or individuals in the VS1 phone system, or to initiate other actions. The Auto Attendant tells the caller what keys on their telephone to press, and then responds to caller input. The usual response is routing a caller to an individual extension, Work Group or voice file location on the VS1 system.

VS1 Auto Attendants enable callers to:

- Access an individual at a specific extension
- Access a group or department at a number of extensions (Work Groups)
- Access an individual or group’s Voice Mail
- Access an operator
- Access recorded messages with information about the company, products, company location, additional routing information, and so on
- Access the main Auto Attendant greeting, and then choose new routing through the system
- Access an extension, Guest Mail Box, or an extension’s Voice Mail using the Dial-by-Name feature
- Record a message
- Hear a phrase
- Hear current date and time
- Run a Command Script

Auto Attendants can be configured to operate for Day (normal hours of operation), Night, Weekend, Holidays, or any other configuration.

The VS1 system has 20 built-in Auto Attendants, three of which are preconfigured and working Auto Attendants. The 17 remaining built-in Auto Attendants can be configured to meet customer requirements. There are also 10 sample Auto Attendants that are provided as examples for learning and practicing purposes. You can configure these samples, but they cannot be used to answer calls. In addition, the first four of these samples are configured as follow:

AA Sample 1 – Sample Auto Attendant for Sales Demos
AA Sample 2 – Sample Auto Attendant for performing Centrex Transfers
AA Sample 3 – Sample Auto Attendant for answering DID lines & transferring the calls to the correct stations automatically
AA Sample 4 – Sample Auto Attendant for Four Digit Extensions

The Four Basic Functions of Auto Attendants

Auto Attendants are designed to anticipate caller actions and respond with an appropriate action. The Auto Attendant performs four basic functions that you customize for each customer. If you keep in mind that you must configure each of these four elements for every Auto Attendant you create, then creating Auto Attendants becomes an easier task. The basic functions include:

Caller Greeting -The Auto Attendant is typically configured to greet callers with a recorded message and provide the caller with information and instructions on how to respond in order to reach an extension, leave a Voice Mail message, or receive additional information.

Get Caller Response - The Auto Attendant is typically configured to recognize a maximum amount of touch-tone digits pressed by the caller within a certain length of time. The number of digits and the time in seconds must be defined for a Caller Response.

Caller Response Handler -The Caller Response Handler consists of two Functions: the **Caller Response** handles caller actions and performs assigned **System Actions**. System Actions include:

- Transfer to Extension
- Transfer to Outside Line
- Dial a Number
- Play Message
- Record Message
- Say Phrase
- Say Current Date & Time
- Run System Command Script

Example - The most basic Auto Attendant would perform the following functions:

- It plays a message to the caller, such as: *"Thank you for calling XYZ, Inc. You may enter the extension number you wish to reach at any time or press 0 to reach the operator."* (Caller Greeting)

- It allows the caller to enter a maximum of four numbers within 3 seconds. (Get Caller Response)

- It accepts a three-digit response (Caller Response) and routes the caller to an extension (System Action).

- It accepts a four-digit response of 6 plus an extension (Caller Response) and routes the caller to that extension's Voice Mail (System Action).

- It accepts a one-digit response of 0 (Caller Response) and routes the caller to the operator at Extension 0 (System Action).

Preconfigured Auto Attendants

The VS1 phone system comes with three ready-to-use Auto Attendants which are utilized by the VS1 phone system. As with any other Auto Attendant script, you can edit and recompile these scripts to modify the way they operate. However, exercise caution when modifying or replacing these files as they are essential functions of the VS1 phone system.

Default Auto Attendant - operates as the main or primary Auto Attendant. Consists of script that handles caller responses and routes callers to the appropriate Extension, Voice Mail, Operator, or Dial by Name Procedure. It is for use with most basic configurations and can be modified if you want to insert voice messages containing information about a specific company.

Before you begin configuring an Auto Attendant, review the Default Auto Attendant (page 95). This will help you when writing an Auto Attendant script. When you review this Auto Attendant, take time to observe the four functions that an Auto Attendant must have in order to function properly.

Voice Message Recorder Auto Attendant- enables you to record and playback messages. Only in unusual circumstances would you modify this script. This and the Phrase List (a listing of

prerecorded words and phrases provided to aid in creating standard messages) are designed for recording voice messages for use by the Auto Attendant.

Voice Mail Interrupt Auto Attendant - gives any caller who has gained access to an extension's Voice Mail the option of entering the Auto Attendant to reach other individuals or groups within the company. It enables callers to choose other options, such as routing to an operator from Voice Mail by pressing 0 before or after a message. ***You need to modify this Auto Attendant to include the end-user business name instead of Telecor, and insert voice messages containing specific information appropriate for the end-user application.***

Note	The VS1 phone system reserves message numbers 11 through 13 and 900 through 999 for your use in creating an Auto Attendant script. *See "Pre-recorded Message List" in Reference section for a list of those messages. In addition, see "Phrase List" in the Reference section, to view a listing of pre-recorded words and phrases to aid in creating standard messages.*

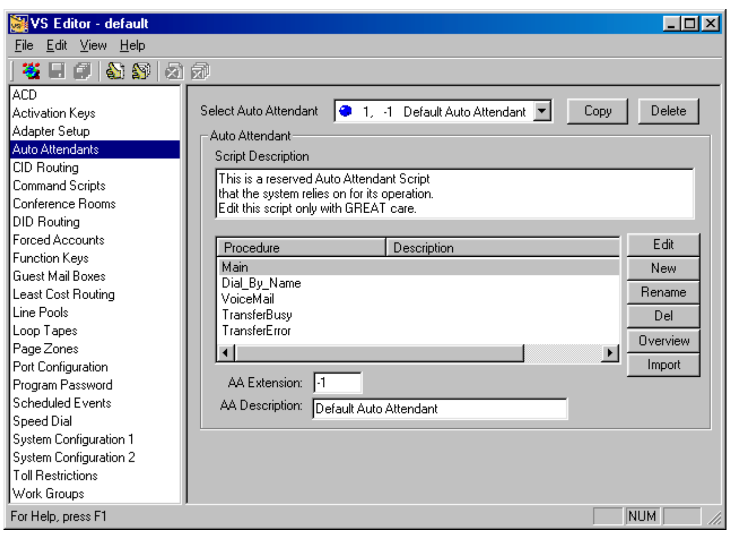
Reviewing the Default Auto Attendant

Before you begin configuring an Auto Attendant, it is recommended you review the default, ready-to-use Auto Attendant. This will help you when writing an Auto Attendant script. Take time to observe the four functions that an Auto Attendant must have in order to function properly. To review the Default Auto Attendant follow these steps:

1. Click **Auto Attendants** in the Tree Control display.

   - The **Auto Attendant** pane appears.

2. In the **Select Auto Attendant** drop-down box, select the **Default Auto Attendant**.



Auto Attendant pane

3. Review the Default Auto Attendant script file. The Default Auto Attendant includes the five Procedures below. A Procedure holds the four basic functions required of an Auto Attendant. There can be multiple Procedures in an Auto Attendant, which allow additional responses from the caller to send him or her to different menus within the same Auto Attendant.

Main: Consists of script that handles caller responses and routes callers to the appropriate Extension, Voice Mail, Operator, or Dial by Name Procedure.

Dial by Name: Consists of the non-accessible system programming that allows the caller to enter the first three letters of a person's name in order to be transferred to that person's extension.

Voice Mail: Consists of script that routes a caller to an extension's voice mail. This Procedure is used when the caller presses 6 with the intention of entering an extension's voice mail, but does not follow it up with an extension number.

TransferBusy: Consists of preconfigured script that handles callers when a busy signal is received at an extension. This Procedure is included by default with every Auto Attendant.

TransferError: Consists of preconfigured script that routes callers back to the Caller Greeting in the event of system errors. This Procedure is included by default with every Auto Attendant.

Click **Overview** to display an overview of the Auto Attendant.

Creating an Auto Attendant

1. Click **Auto Attendants** in the Tree Control display.

2. The **Auto Attendant** pane appears. From the **Select Auto Attendant** drop-down box select an empty Auto Attendant. The first three Auto Attendants are Preconfigured Auto Attendants, which you can edit and modify. The drop-down box also includes 10 sample Auto Attendants that appear as you scroll down. You can configure these samples, but they cannot be used to answer calls. The blue circle adjacent to an Auto Attendant indicates that it has been configured.

3. Select an empty **Auto Attendant.**

4. In the **Script Description** text box, enter a brief description about the purpose of the Auto Attendant.

5. The **Procedure Description** list box is where the Procedures are listed. The **Main** Procedure consists of the script that is integral to the functioning of the Auto Attendant, and is the one that you will modify. The **TransferBusy** Procedure is preconfigured and handles callers when a busy signal is received at an extension. The **TransferError** Procedure is preconfigured and routes callers back to the Caller Greeting in the event of system errors.

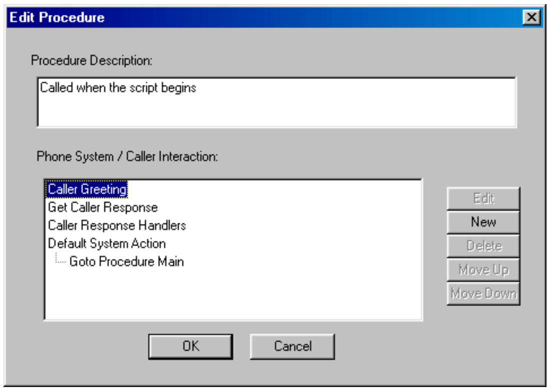
6. In the **AA Extension** text box, enter an extension for the Auto Attendant.

7. In the **AA Description** text box, enter a name for the Auto Attendant.

8. Double-click the **Main** Procedure (or select it and click **Edit**).

   - The **Edit Procedure "Main"** window appears.

9. In the **Procedure Description** text box, enter a brief description about the purpose of the Procedure.



Edit Procedure window

10. You are now ready to set up the first of the four basic functions: **Caller Greeting**.

Caller Greeting

In the **Phone System / Caller Interaction** list box, select **Caller Greeting** and click **New**. The **New Caller Greeting** window appears with the following options:

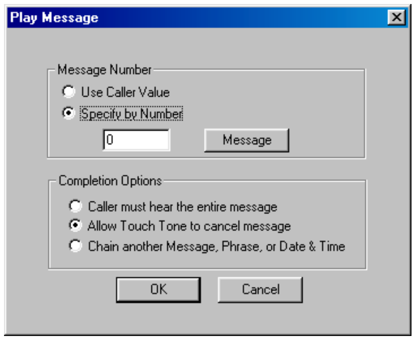
Play Message: Plays a recorded voice message.
Say Phrase: Plays a series of pre-defined words assembled to produce a short message.
Say Current Date & Time: Plays the current date and time.
Run System Command File: Runs a System Command.

New Caller Greeting window

Caller Greeting – Play Message

Clicking **Play Message** in the **New Caller Greeting** dialog box brings up the **Play Message** dialog box. Complete the following steps:

1. In the **Message Number** group box, select **Specify by Number** to play a pre-selected voice message.

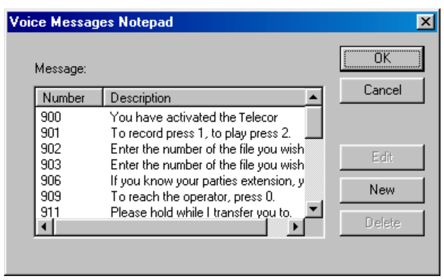


Play Message dialog box

2. Click in the **Specify by Number** text box that appears and type in a voice file number. Note that voice file numbers 11-13 and 900-999 are already reserved with voice files. *See "Pre-recorded Message List" in Reference section for a list of reserved messages.*

3. Click the **Message** button.

4. The **Voice Message Notepad** window is displayed. Click **New** to describe the voice file you typed in the **Specify by number** text box. A message needs to be assigned a number and its text content. The actual message is stored by the Voice Message Recorder Auto Attendant. Message files for all Auto Attendants appear in this window. Use care when deleting or editing one of these messages because all Auto Attendants using that message are affected. *See "Recording a Voice File" in Reference section to record a voice file.*



Voice Messages notepad dialog box

5. Select a **Completion Option.**

- **Caller Must Hear the Entire Message:** This option is used when a Voice Message is being played and the caller is not allowed to interrupt the message with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Voice Message continues playing until it is complete.

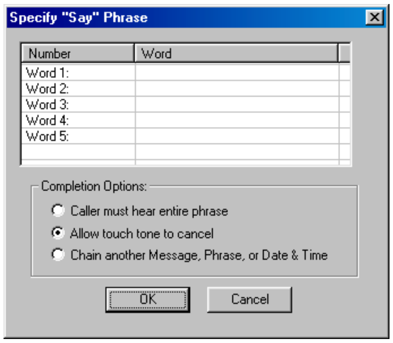
- **Allow Touch Tone to Cancel Message:** Allow Touch Tone to Cancel Message is the most common option selected. With this option, the caller may interrupt the playing of the Voice Message with a DTMF tone. The Auto Attendant immediately takes action on the tone(s) entered by the caller.

Note If you are playing a series of consecutive voice files (Messages, Phrases, Date & Time) within one Caller Greeting, do not choose this option for any of the voice files except the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

- **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) within one Caller Greeting. The Auto Attendant begins to play the file and immediately moves on to the next voice file. An example of proper usage: Four voice files are stacked for consecutive play.

Caller Greeting – Say Phrase

Clicking **Say Phrase** in the **New Caller Greeting** dialog box brings up the **Specify "Say" Phrase** dialog box.



Specify "Say" Phrase dialog box

1. For the first word in the phrase, click in the space adjacent to **Word 1** under the **Word** column.

2. A drop-down box appears. Select a word from the Phrase List. *See "Phrase List" in Reference section for a list of words.*

3. Repeat above steps for additional words in the phrase. A maximum of 5 words can be chosen for a phrase.

4. Select a Completion Option.

- **Caller Must Hear Entire Phrase:** This option is used when the Phrase is being played and the caller is not allowed to interrupt it with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Phrase continues playing until it is complete.

- **Allow Touch Tone to Cancel:** Allow Touch Tone to Cancel is the most common option selected. With this option, the caller may interrupt the playing of the Phrase with a DTMF tone. The Auto Attendant immediately takes action on the tone(s) entered by the caller.

Note If you are playing a series of consecutive voice files (Message, Phrase, Date & Time) within one Caller Greeting, *do not* choose this option for any of the voice files *except* the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

- **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) within one Caller Greeting. The Auto Attendant begins to play the file and immediately moves on to the next voice file. An example of proper usage: Four voice files are stacked for consecutive play.

Caller Greeting – Say Current Date & Time

Clicking **Say Current Date and Time** in the **New Caller Greeting** dialog box brings up the **Say Current Date and Time** dialog box. You must choose a Completion Option.

Say Current Date and Time dialog box

- **Caller Must Hear Entire Date and Time:** This option is used when the Date and Time are being played and the caller is not allowed to interrupt it with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Date and Time continue playing until complete.

- **Allow Touch Tone to Cancel:** Allow Touch Tone to Cancel is the most common option selected. With this option, the caller may interrupt the playing of the Date and Time with a DTMF tone. The Auto Attendant immediately takes action on the tone(s) entered by the caller.

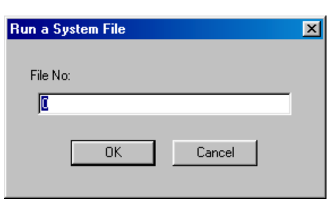
Note If you are playing a series of consecutive voice files (Message, Phrase, Date & Time) within a Caller Greeting, *do not* choose this option for any of the voice files *except* the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

- **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) within a Caller Greeting. The Auto Attendant begins to play the file and immediately moves on to the next file. An example of proper usage: Four voice files are stacked for consecutive play.

Caller Greeting – Run System Command File

Clicking **Run System Command File** in the **New Caller Greeting** dialog box brings up the **Run a System File** dialog box.

1. In the **File No.** text box, enter the two-digit Command Script number. *See Command Scripts on page 116 for information on creating a Command Script.*



Run a System File dialog box

Get Caller Response

Once the Caller Greeting is set up, the Procedure must be configured to receive and recognize touch-tone caller responses to instructions given during the Caller Greeting. The maximum number of digits and a timeout must be defined in the **Get Caller Response**.

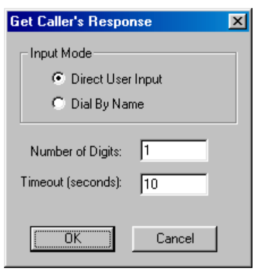
1. In the **Phone System / Caller Interaction** list box, select **Get Caller Response** and click **New**.

    - The **Get Caller's Response** dialog box appears.

2. In the **Input Mode** group box, select **Direct User Input**.

3. In the **Number of Digits** text box, enter the maximum amount of digits the caller can enter.

4. In the **Timeout (seconds)** text box, enter a number (representing seconds) that determines how long the system waits before acting if a caller does not respond. The Timeout timer starts after the greeting is played.



Get Caller Response dialog box

Caller Response Handler

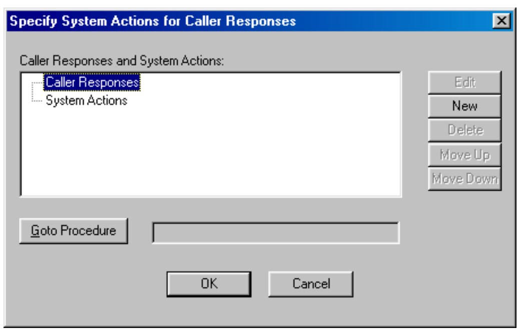
The Procedure must be configured to handle caller responses in order to activate a System Action or route callers to another Procedure. The **Caller Response Handler** consists of two Functions: the **Caller Response** handles caller actions and performs assigned **System Actions**. System Actions include:

- Transfer to Extension
- Transfer to Outside Line
- Dial a Number
- Play Message
- Record Message
- Say Phrase
- Say Current Date & Time
- Run System Command Script

Caller Response

1. In the **Phone System / Caller Interaction** list box, select **Caller Response Handlers** and click **New**.

   - The **Specify System Actions for Caller Responses** dialog box appears.



Specify System Actions for Caller Responses dialog box

2. Double-click **Caller Responses** (or select it and click **New**).

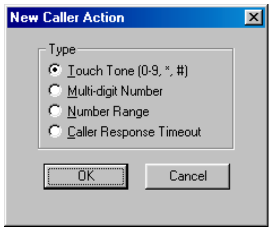
   - The **New Caller Action** dialog box appears with the following options:

     Touch Tone (0-9, *, #): Select this option to have the caller press a single keypad number.

     Multi-digit Number: Select this option to have the caller press a series of multiple keypad numbers.

     Number Range: Select this option to have the caller target a number(s) within a set range.

     Caller Response Timeout: Select this option to choose a System Action or route the caller to another Procedure if the caller does not respond within the time determined in the Get Caller Response.



New Caller Action dialog box

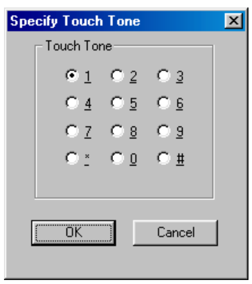
Caller Response – Touch Tone

Clicking **Touch Tone (0-9, *, #)** in the **New Caller Action** dialog box brings up the **Specify Touch Tone** dialog box.

1. Choose a number or symbol that the caller is required to press.

2. Click **OK**.

3. Two options are available: Send the Caller to another Procedure (page 104) or Assign a System Action (page 105).



Specify Touch Tone dialog box

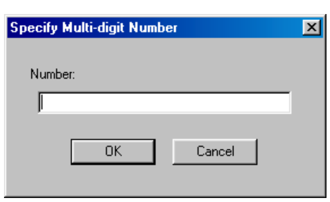
Caller Response – Multi-digit Number

Clicking **Multi-digit Number** in the **New Caller Action** dialog box brings up the **Specify Multi-digit Number** dialog box.

1. In the **Number:** text box, enter a series of numbers that the caller is required to press (maximum 10 digits).

2. Click **OK**.

3. Two options are available: Send the Caller to another Procedure (page 104) or Assign a System Action (page 105).



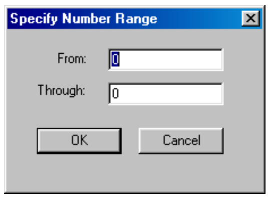
Specify Multi-digit Number dialog box

Caller Response – Number Range

Clicking **Number Range** in the **New Caller Action** dialog box brings up the **Specify Number Range** dialog box.

1. Type a valid range using the **From** and **Thru** text boxes.
2. Click **OK**.
3. Two options are available: Send the Caller to another Procedure (see below) or Assign a System Action (page 105).

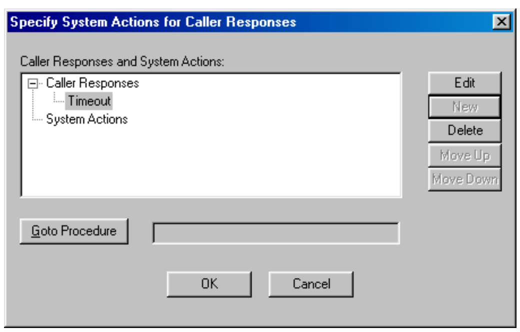
Specify Number Range dialog box



Caller Response – Caller Response Timeout

If the caller does not respond within the time allocated in the **Get Caller Response**, two options are available: Send the Caller to another Procedure (see below) or Assign a System Action (page 105).

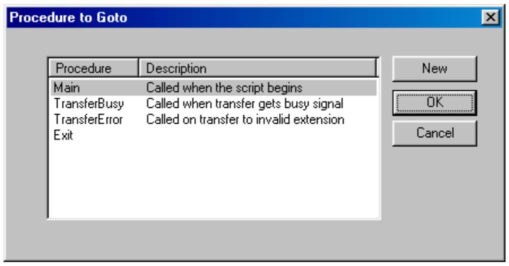
Timeout selected for Caller Response



Sending the Caller to Another Procedure

After creating a Caller Response, you are returned to the **Specify System Actions for Caller Responses** window. To send the caller to another procedure:

1. Click **Goto Procedure**.
   - The **Procedure to Goto** dialog box appears.



Procedure to Goto dialog box

2. Double-click the desired Procedure.

   - The Procedure appears beside the **Goto Procedure** button.

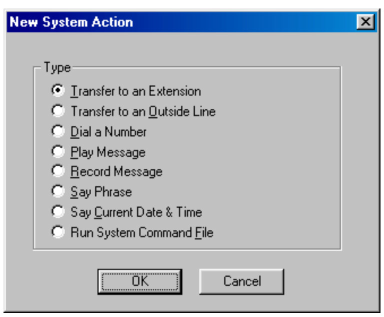
Note Selecting the **Exit** Procedure disconnects the call.

Assigning a System Action

After creating a Caller Response, you are returned to the **Specify System Actions for Caller Responses** window. To assign a system action:

1. Select **System Actions** and click **New**.

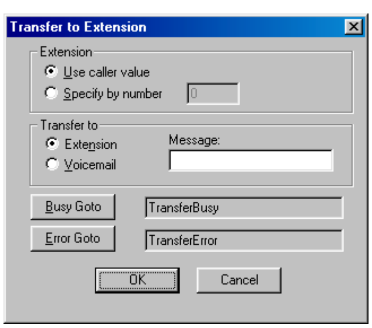
   - The **New System Action** dialog box appears displaying the options shown below:



New System Action dialog box

System Action – Transfer to an Extension

Clicking **Transfer to Extension** in the **New System** dialog box brings up the **Transfer to Extension** dialog box.



Transfer to Extension dialog box

1. In the **Extension** group box, select one of the following options:

   Use Caller Value: the caller will be transferred to a number entered by caller in the Caller Response.
   Specify by Number: the caller is transferred to the number entered in the **Specify by Number** text box.

2. In the Transfer To group box, select one of the following two options:

   Extension: transfers the caller directly to the extension.
   Voice Mail: transfers the caller to the extension's Voice Mail.

3. You can add messages in the **Message** text box. Messages can contain up to 16 characters. The message appears on Display Phones and CTI Stations. The message is also stored in SMDR. This feature is especially useful for call screening and for use with DID numbers, as used by answering services.

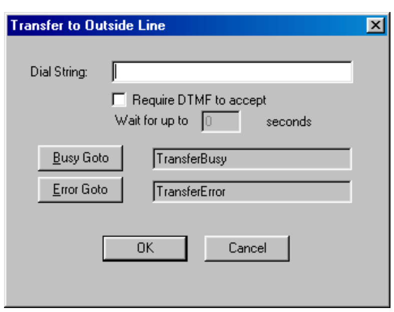
4. The **TransferBusy** procedure is pre-selected for **Busy Goto**. This procedure handles callers when a busy signal is received at an extension. The **TransferErro**r procedure is pre-selected for **Error Goto**. This procedures routes callers back to the Caller Greeting in the event of system errors. These procedures are preconfigured and are included in every Auto Attendant.

5. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

6. A Goto Procedure value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** window appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action – Transfer to an Outside Line

Clicking **Transfer to an Outside Line** in the **New System** dialog box brings up the **Transfer to an Outside Line** dialog box.



Transfer to Outside Line dialog box

1. In the **Dial String** text box, enter phone number of the outside line that the caller will be transferred to.

2. Check **Require DTMF to Accept** in order to have the VS1 system play the following message when the called party answers: "You have a call from the VS1 System. Press 1 to accept or simply hang up." If this option is chosen, you must enter a time in seconds in the **Wait for up to ___ seconds**. If the called party does not press 1 within this time, the call is not accepted.

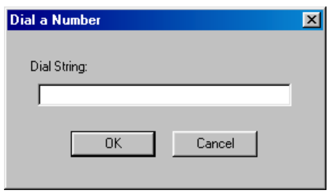
3. The **TransferBusy** procedure is pre-selected for **Busy Goto**. This procedure handles callers when a busy signal is received at the called party. The **TransferError** procedure is pre-selected for **Error Goto**. This procedure routes callers back to the Caller Greeting in the event of system errors. These procedures are preconfigured and are included in every Auto Attendant.

4. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

5. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** window appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action – Dial a Number

Clicking **Dial a Number** in the **New System** dialog box brings up the **Dial a Number** dialog box.



Dial a Number dialog box

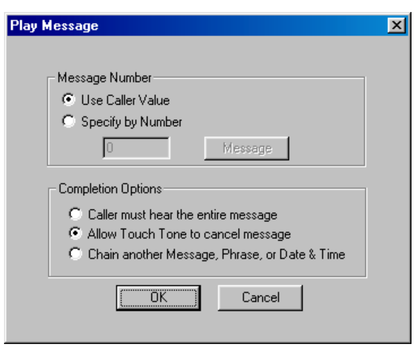
1. In the **Dial String** text box, enter a string of digits to be dialed. Valid characters include a comma (**,**) for a pause and **F** for external flash.

2. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

Note	The **Dial a Number** System Action is mainly used to perform Centrex Transfers. View Sample Auto Attendant 2 for an example.

System Action – Play Message

Clicking **Play Message** in the **New System** dialog box brings up the **Play Message** dialog box.



Play Message dialog box

1. In the **Message Number** option box, select one of the following two options:

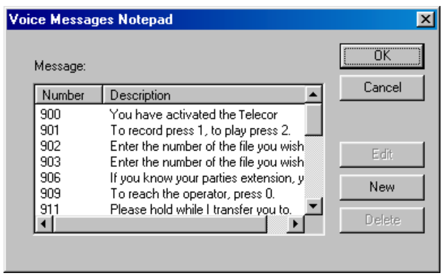
   Use Caller Value: plays the voice message number that was previously dialed by caller in Caller Response.

Specify by Number: plays a pre-selected voice message number

2. If **Specify by Number** is selected, click in the **Specify by Number** text box that appears and type in a voice file number. Note that voice file numbers 11-13 and 900-999 are already reserved with existing voice files. *See "Pre-recorded Message List" in Reference section for a list of reserved messages.*

3. Click the **Message** button.

   - The **Voice Message Notepad** window is displayed. Use this window to describe the voice file you typed in the **Specify by number** text box. A message is assigned a number and brief description. The actual message is stored by the Voice Message Recorder Auto Attendant. Message files for all Auto Attendants appear in this window. Use care when deleting or editing one of these messages because all Auto Attendants using that message are affected. *See "Recording a Voice File" in Reference section to record a voice file.*



Voice Messages notepad dialog box

4. Select a **Completion Option.**

   - **Caller Must Hear the Entire Message:** This option is used when a Voice Message is being played and the caller is not allowed to interrupt the message with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Voice Message continues playing until it is complete.

   - **Allow Touch Tone to Cancel Message:** Allow Touch Tone to Cancel Message is the most common option selected. With this option, the caller may interrupt the playing of the Voice Message with a DTMF tone. The Auto Attendant then skips the Voice Message and moves onto the next System Action or the Goto Procedure.

Note If you are playing a series of consecutive voice files (Messages, Phrases, Date & Time) within one System Action, do not choose this option for any of the voice files except the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

   - **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) in a System Action. The Auto Attendant begins to play the file and immediately moves on to the next voice file. An example of proper usage: Four voice files are stacked for consecutive play.

5. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

6. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action – Record Message

Clicking **Record Message** in the **New System** dialog box brings up the **Record Message** dialog box.

Record Message dialog box

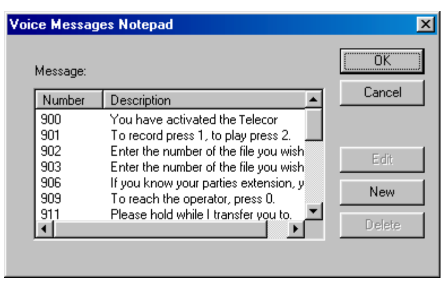
1. Select one of two options:

   Use Caller Value: will record a caller voice file and save voice file into number specified by caller in Caller Response.
   Specify by Number: will record a caller voice file and save voice file into number specified in accompanying text box.

2. If **Specify by Number** is selected, click in the **Specify by Number** text box that appears and type in a voice file number. Note that voice file numbers 11-13 and 900-999 are already reserved with existing voice files. *See "Pre-recorded Message List" in Reference section for a list of reserved messages.*

3. Click the **Message** button.

   - The **Voice Message Notepad** window is displayed. Use this window to describe the voice file you typed in the **Specify by number** text box. A message is assigned a number and brief description. The actual message is stored by the Voice Message Recorder Auto Attendant. Message files for all Auto Attendants appear in this window. Use care when deleting or editing one of these messages because all Auto Attendants using that message are affected.



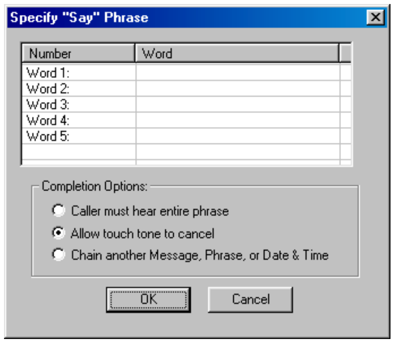
Voice Messages notepad dialog box

4. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

5. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action – Say Phrase

Clicking **Say Phrase** in the **New System** dialog box brings up the **Specify "Say" Phrase** dialog box.



Specify "Say" Phrase dialog box

1. Click in the space adjacent to **Word 1**. Click **Lookup Words**.

2. The **Phrase List** appears. Choose a word to be first in the phrase.

3. Repeat above steps to complete your phrase.

4. Select a Completion Option.

- **Caller Must Hear Entire Phrase:** This option is used when the Phrase is being played and the caller is not allowed to interrupt the it with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Phrase continues playing until it is complete.

- **Allow Touch Tone to Cancel:** Allow Touch Tone to Cancel is the most common option selected. With this option, the caller may interrupt the playing of the Phrase with a DTMF tone. The Auto Attendant then skips the Phrase and moves onto the next System Action or the Goto Procedure.

Note If you are playing a series of consecutive voice files (Message, Phrase, Date & Time) within one System Action, *do not* choose this option for any of the voice files *except* the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

- **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) within one System Action. The Auto Attendant begins to play the file and immediately moves on to the next voice file. An example of proper usage: Four voice files are stacked for consecutive play.

5. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

6. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action – Say Current Date and Time

Clicking **Say Current Date & Time** in the **New System** dialog box brings up the **Say Current Date and Time** dialog box.

Say Current Date and Time dialog box

1. Choose a Completion Option.

- **Caller Must Hear Entire Date and Time:** This option is used when the Date and Time are being played and the caller is not allowed to interrupt the it with DTMF tones. If the caller does enter a tone, it is ignored by the system and the Date and Time continue playing until complete.

- **Allow Touch Tone to Cancel:** Allow Touch Tone to Cancel is the most common option selected. With this option, the caller may interrupt the playing of the Date and Time with a DTMF tone. The Auto Attendant then skips the Date and Time and moves onto the next System Action or the Goto Procedure.

Note If you are playing a series of consecutive voice files (Message, Phrase, Date & Time) within System Action, *do not* choose this option for any of the voice files *except* the last one in the series. If you set up all the files in the series to use this option, the Auto Attendant recognizes the DTMF tone(s) only for the purpose of skipping to the next voice file in the series, not for the purpose of taking action on the caller's response.

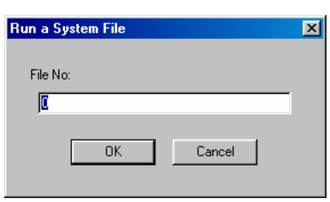
- **Chain another Message, Phrase or Date & Time:** This option is used, typically, for only one purpose—stacking a series of voice files (Message, Phrase, Date & Time) within one System Action. The Auto Attendant begins to play the file and immediately moves on to the next file. An example of proper usage: Four voice files are stacked for consecutive play.

2. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

3. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

System Action - Run System Command Script

Clicking **Run System Command File** in the **New System** dialog box brings up the **Run a System File** dialog box.



Run a System File dialog box

1. In the **File No.** text box, enter the two-digit Command Script number. *See Command Scripts on page 116 for information on creating a Command Script.*

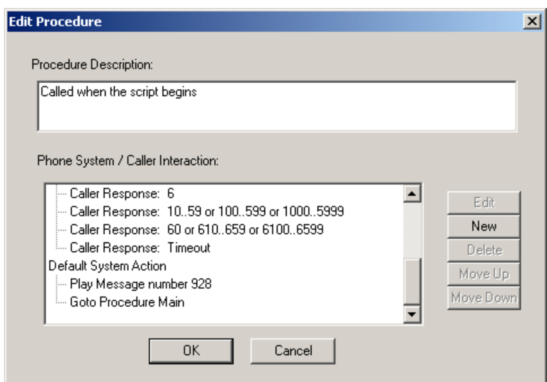
2. Click **OK** to return to the **Specify System Actions for Caller Responses** dialog box.

3. A **Goto Procedure** value is required, even though it may not be used. Click **Goto Procedure**. The **Procedure to Goto** dialog box appears. Double-click a procedure to select it (the **Main** Procedure is the most commonly used choice).

Default System Action

By default, each Procedure is provided with a Default System Action. This Default System Action returns the caller back to the Main Procedure if an invalid Caller Response is entered. In addition, the Main Procedure and Voice Mail Procedure of AA #1 (Default AA) and AA #3 (Voice Mail Interrupt AA) plays message 928, which states "I'm sorry, that is not a valid option."

Default System Action of Main Procedure of AA#1



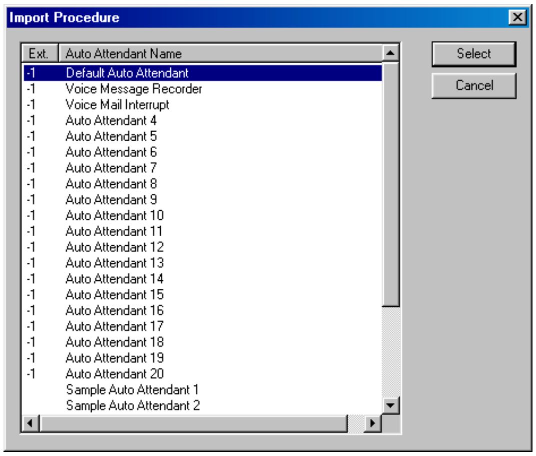
Importing an Auto Attendant Procedure

You can import Procedures of Auto Attendants for use in creating new Auto Attendants. For example, the Dial by Name Procedure of Auto Attendant 1 can be imported into a new Auto Attendant.

1. From the **Select Auto Attendant** drop-down box, select an Auto Attendant that will receive an imported Procedure.

2. Click **Import**.

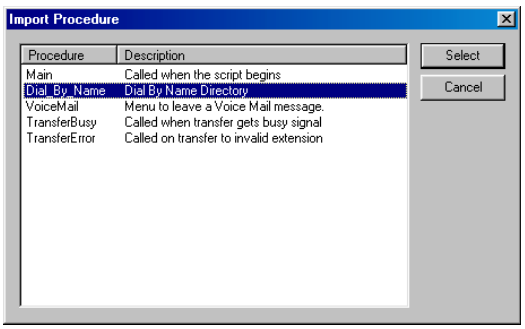
- The **Import Procedure** dialog box appears with a list of Auto Attendants.



Import Procedure dialog box displaying Auto Attendants

3. Double-click the Auto Attendant whose Procedure will be imported.

    - The **Import Procedure** dialog box displays the Procedures for the Auto Attendant.



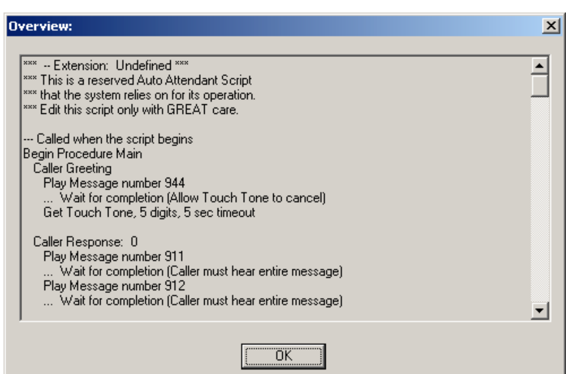
Import Procedure dialog box displaying Auto Attendant Procedures

4. Double-click the Procedure to be imported.

    - The Procedure is imported into the Auto Attendant.

Viewing the Auto Attendant Script

The Auto Attendant script can be viewed in its entirety by clicking the **Overview** button.



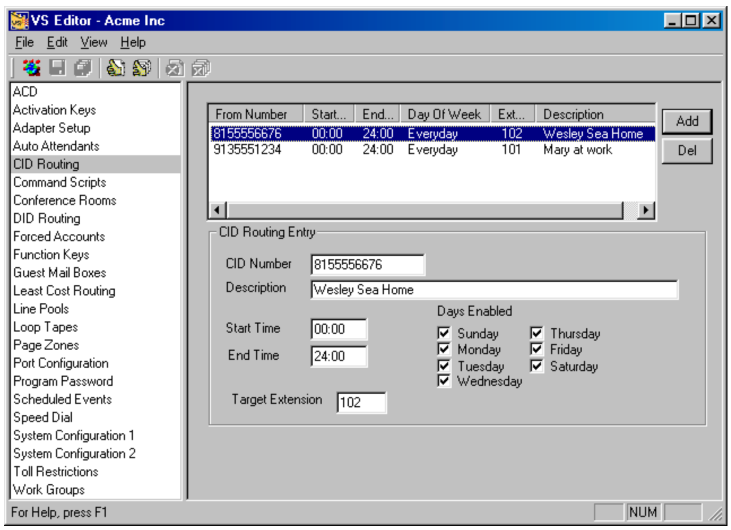
Overview of Auto Attendant Script

CALLER ID ROUTING

Caller ID Routing enables incoming calls to be targeted to specific extensions based on Caller ID. Each time a call is received from that specific telephone number, the call is automatically routed to a preassigned extension. You can set up Caller ID Routing for up to 800 entries.

For example, your spouse calls your company from (913) 555-1234. The number 9135551234 was set up in the Caller ID Routing Entry table to target extension 101 with the description, *Mary at work*. Any time a call is placed from (913) 555-1234 to the company, extension 101 receives that call with the Caller ID description, *Mary at work*.

Note You must contact your local phone company to activate Caller ID, or Caller ID Routing will not work. If you are using Port Expansion Unit Models 205 and 200, the Caller ID Option Module must be installed.



CID Routing pane

Complete the following steps:

1. Select **CID Routing** in the Tree Control display.

   - The **CID Routing** pane appears.

2. Click **ADD**.

3. In the **CID Routing Entry** group box, enter the following in the corresponding text boxes:

   CID Number: Enter the number of the incoming call you want to specify. Do not use dashes, spaces, or parenthesis when entering the number.

Note You must include the area code even if it is a local call, or Caller ID Routing will not function properly. Wildcards can be used in the CID Routing Entry table. Use **x** for multiple numbers and **?** for single digits.

   Description: Type a description (maximum 16 characters) for the specified incoming number.

Start Time and **End Time**: Set a time frame for the Caller ID number you specified. Times must be entered in standard 24-hour (military) format. If you want a call routed anytime, type **00:00** in the **Start Time** text box and type **24:00** in the **End Time** text box.

Note	The **End Time** must not come before the **Start Time**.

Days Enabled: Check the day(s) of the week that the incoming number will be routed.

Target Extension: Enter the extension that you want the incoming call routed to. You can only type one extension. You can target an incoming call to a Work Group extension. By doing this, all phones in the specified Work Group ring and display the specified description.

4. Click the **Save** button in the toolbar.
    - The CID Routing entry is added to the list box above.
5. Repeat steps 2-4 for other CID Routing entries.

To edit a Caller ID Routing Entry, select it from the list box and make the required edits.

To delete a Caller ID Routing Entry, select it from the list box and click **Del**. A dialog box appears asking you to confirm your deletion. Click **Yes** and your entry is deleted.

Routing Calls from a Specific Area Code

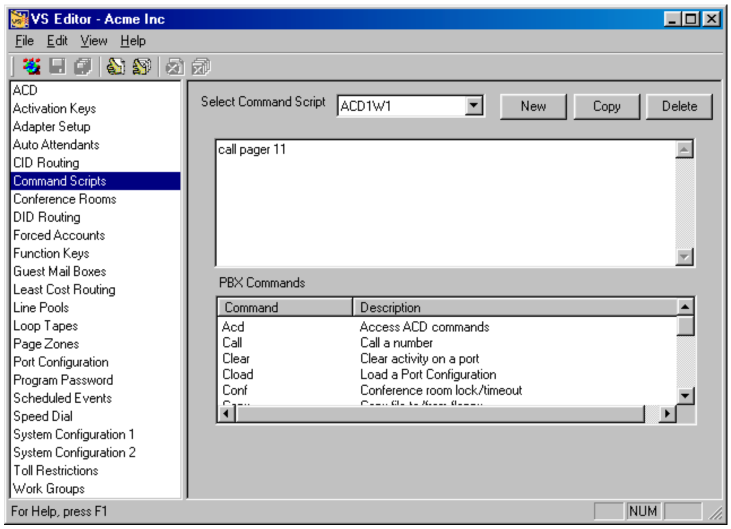
Caller ID Routing can be set up to transfer incoming calls from a specific area code to a preassigned extension. Complete the following steps:

1. Select **CID Routing** from the Tree Control display.
    - The **CID Routing** pane appears.
2. In the **CID Number** text box type in the area code followed by the wildcard **x**.
3. Fill in the **Description, Start Time**, **End Time**, **Days**, and **Target Extension** boxes.

COMMAND SCRIPTS

Command Scripts are batches of VS1 System Commands. Command Scripts provide a convenient method for controlling the various aspects of system operation. Command scripts can be run from:

- The TVS Command prompt
- A VS1 System phone
- From an off-site location via an Auto Attendant



Command Scripts pane

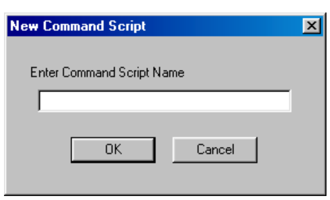
Complete the following steps:

1. Select **Command Scripts** in the Tree Control display.

   - The **Command Scripts** pane appears.

2. Click **New** to create a new Command Script.

3. In the **New Command Script** dialog box, type a name in the **Command Script Name** text box and click **OK**.



New Command Script dialog box

Note If you are running Command Scripts from the TVS Command prompt, the Command Script can be named using numbers and/or letters. However, if you also plan on running Command Scripts from system phones or an Auto Attendant, you must name the Command Script using the format of **f{nn}** where **{nn}** is a number in the range of 1–99. *For more information, see “Running a Command Script from an Auto Attendant” on page 118.*

4. System Commands are located in alphabetical order in the **PBX Commands** list box. Double-click the System Command to include it newly-created Command Script.

5. Click the **Save** button in the toolbar.

Default Command Scripts

The VS1 includes default Command Scripts, which are explained below. View these Command Scripts in the VS1 Editor to see what System Commands are used.

ACD1W1 – Plays voice file 011 ("There are too many calls for the number of agents. Please add another agent.")

ACD1W2 – Plays voice file 012 ("Calls in queue have been waiting too long. Please add another agent.")

ACD1W3 – Plays voice file 013 ("Calls are in queue and zero agents are logged in. Please add agents.")

DAY – Loads the day configuration

DBNFIRST – Configures Dial-by-Name for first name.

DBNLAST – Configures Dial-by-Name for last name.

DSKMAINT – Triggers the run of disk maintenance utilities *(see "Scheduled Disk Optimization" in Reference section*)

F1 – Manually loads the DAY configuration

F2 – Manually loads the NIGHT configuration

F93 – Forces the RSA modem off-hook in answer mode

F95 – Resets the RSA modem

NIGHT – Loads the NIGHT configuration

SMDRMON – Backs up SMDR data to summary.mon, detail.mon, and acd.mon files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRTUE – Backs up SMDR data to summary.tue, detail.tue, and acd.tue files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRWED – Backs up SMDR data summary.wed, detail.wed, and acd.wed files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRTHU – Backs up SMDR data to summary.thu, detail.thu, and acd.thu files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRFRI – Backs up SMDR data to summary.fri, detail.fri, and acd.fri files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRSAT – Backs up SMDR data to summary.sat, detail.sat, and acd.sat files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRSUN – Backs up SMDR data to summary.sun, detail.sun, and acd.sun files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

STARTUP - Is executed every time the TVS software starts. Commands that are used to modify default system settings should be placed in this Command Script.

Running a Command Script from the TVS Command Prompt

1. To run a Command Script from the TVS Command prompt use the format: **`@{file}`** (with `{file}` representing the Command Script Name/Number).

2. Press ENTER.

Running a Command Script from a Phone

1. Lift the handset, and dial **71** and the two-digit Command Script Number that you set up to run a command.

   - For example: dial **7101** to run the **`f1.cmd`**; dial **7102** to run the **`f2.cmd`**; dial **7199** to run the **`f99.cmd`**.

Note	You can customize a Feature button on the Display Phone Model 200 (DP200) to run a Command Script. Lift the handset and dial **7801**, and then press **2** to customize a Feature button. Dial **32** for the Command Script code number, and then dial the two digit Command Script that you set up to run.

Running a Command Script from an Auto Attendant

Running Command Scripts from an Auto Attendant provides customers the flexibility of being in control of how their system operates even when they are not on-site. Following the steps below ensures that authorized personnel have the option of calling in to the VS1 phone system and dialing the Command Script Number when the Auto Attendant answers.

1. Click **Auto Attendants** in the Tree Control display.

   - The **Auto Attendant** pane appears.

2. From the **Select Auto Attendant** drop-down box, select the Auto Attendant to run the Command Script.

3. Select the **Main** procedure, and then click **Edit**.

4. In the **Phone System / Caller Interaction** list box, select **Caller Response Handlers** and click **New**.

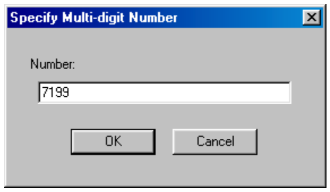
   - The **Specify Systems Actions for Caller Responses** dialog box appears.

5. Select **Caller Responses** and click **New**.

6. In the **New Caller Action** dialog box, select **Multi-digit Number**.

7. In the **Specify Multi-digit Number** window, type **71{nn}** (with **nn** representing the Command Script number) in the **Number** text box and click **OK**.

   - Placing **71** prior to the Command Script number ensures that the code to run the Command Script is the same externally as it is internally (running Command Script from system Phone). Up to 10 digits can be entered in the **Number** box; however, the code to run the Command Script externally will then be different from running the Command Script internally.

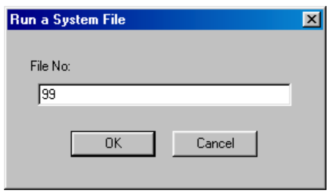


Specify Multi-digit Number dialog box

8. Select **System Actions** and click **New**.

9. Under **Type** in the **New System Action** window, select **Run System Command File**. Click **OK**.

10. In the **File No:** text box in the **Run a System File** window, type the two-digit Command Script number. Click **OK**.



Run a System File dialog box

11. Select **Goto Procedure** and then click **Main**. Click **OK** to save.

Note The above steps use the **Multi-digit Number** Caller Response to run the Command Script. This is the preferred method. However, the other three methods of Caller Response (Touch Tone, Number Range, and Caller Response Timeout) can be also be used if need be. *See "Caller Response Handler" on page 102 in Auto Attendant section for more information*.

System Commands

argument	Indicates an argument for which you must supply a value.
{x}	An argument or a constant within braces { } is required.
[x]	An argument or a constant within square brackets [ ] is optional.
x|y|z	Constants or arguments separated by a vertical bars requires a choice.

acd logon {*port*} {*logon code*}—Logs the specified port onto an ACD(s) using the logon account code specified.

acd logoff {*port*}—Logs the specified port off of all ACD(s).

call ext {*extension*} {*sysfile*}—Plays a voice file at an extension port. Example: **call ext 138 44** would call extension **138** and play voice file **$va044.pcm** at that extension.

call pager {*sysfile*} {*pagezone*}—Plays a voice file over a Paging Zone. Example: **call pager 22 4** would play voice file **$va022.pcm** over Paging Zone 4 .

call remote {*number*} {*sysfile*} [*dtmf*]—Dials a remote number and plays a voice file, optionally dials DTMF. For example, **call remote 8887936 937 1234** would dial **8887936** (using the Outbound Paging Line Pool specified in System Config 1), pause 10 seconds, play voice file **$va937.pcm**, and then dial the DTMF 1234 and hang up.

clear {*port*}—Clears the specified port. Clearing a CO port disconnects the current call. Clearing an Extension port disconnects the current call and cancels any Forwarding or Do Not Disturb that may be in effect.

cload {*file*}—Loads a configuration file. For example, **cload night.cfg** would load the configuration file **night.cfg**.

conf {co|ext} [*input*] [*output*]—Sets the 16-party conference room volume. Contact Telecor Technical Support for further information.

conf lock {0|1}—Controls the ability to lock a 16-party conference room. Locking is accomplished by any party pressing **#** . Pressing ***** unlocks the conference room. A locked conference room results in a busy signal when its extension is dialed. (Default = 1, locking/unlocking enabled.) Typing **conf lock** returns the current value.

conf timeout [*n*]—Controls how long a conference call comprised of only CO ports can continue before the VS1 phone system terminates the call. Setting a value of **0** disables the timer. Pressing a DTMF key resets the timer. (Valid range = 0–32767 minutes, default = 15 minutes) Typing **conf timeout** returns the current value.

copy {*source*} {*destination*}—Copies the file specified as the source to the target. The full path and file name of both the source and target files must be specified. Wildcards are not supported. If the target file exists, it is overwritten. To copy files to/from the network, specify **net:** as the path.

date [*mmddyyhhmm*]—Sets the system date and time using the format **mmddyyhhmm**. Typing **date** by itself displays the current system date and time.

del {*filename*}—Deletes the specified file. Wildcards are not supported, and files with the read-only attribute set cannot be deleted.

dir {*filename*}—Displays a directory of files. Directory path names and wildcards can be specified. For example, **dir vm\vm*.*** would display all files in the Voice Mail storage directory.

disable {*port*|all}—Disabling a CO port prevents it from being used for outbound calls but does not prevent incoming calls. Extension ports should not be disabled.

`dsp enable`—This command must be issued at least once after PBX startup to enable DSP volume adjustments. It is good practice to issue this command at the beginning of any Command Script that performs DSP volume adjustments.

Note	The dsp enable command must be issued before any dsp vol command is acted upon by the system.

`dsp vol {co|ext|`*`port`*`} ={`*`n`*`} {in|out}`—This command allows CO and Extension port volume adjustments through the Host Adapter DSP. Each port (T1 or PEU) has two settings associated with it: one controls the volume level IN to the port and the other controls the volume OUT of the port. For example, **`dsp vol co =5 in`** would change the incoming volume on every CO port to 5 dB. (Valid range = -12 to 12 dB, default is 6 dB for CO IN on analog ports, 0 dB for CO IN on T1 ports, 0 dB for CO OUT, -6 dB for EXT OUT, and 0 dB for EXT IN)

`dump call`—Displays the current status of the paging queue. This queue contains Auto Pages and Pager Notifications that are currently being dispatched.

`dump page`—Displays the current status of the Pager Notification queue. This queue contains Pager Notifications that are scheduled for later release.

`enable {`*`port`*`|all}`—Enables a port previously disabled.

`err`—Displays the current status of the Switch Card and Host Adapter Card(s).

`exit`—This command shuts down the PBX software and reboots the TVS. Can only be issued from local console (monitor and keyboard connected to TVS) and not from an RSA session.

`grpid [off|on|0|1]`—Controls the operation of the Answer Group function. When GRPID is ON, Answer Group picks up calls within the same group only. When GRPID is OFF, Answer Group picks up calls system-wide. Typing GRPID by itself displays the current setting. (Default = ON/1)

`help`—Displays a list of some of the available commands.

`ls [acd|all|co|ext]`—Displays a line status of ports. The **`ls acd`** command displays a summary of configured ACD groups. **`ls co`** and **`ls ext`** displays a list of CO or Extensions, respectively. The **`ls all`** command lists all ports, including Paging Zones, Auto Attendants, Park Zones, and so on. For more information, see “Line Status Symbols,” in Installation, Configuration, and Operating Guide.

`modem answer`—This command causes the TVS modem to go off-hook in answer mode, similar to issuing **`modem send ata`**.

`modem reset 1`—Resets the TVS modem and forces it on-hook, terminating the current RSA session.

`modem send {`*`command`*`}`—Sends the specified command string to the TVS modem in command mode. Be sure to include **`at`** at the beginning of the string.

`pause [`*`n`*`]`—Pauses the current command script for {n} seconds. The **`pause`** command by itself creates a 1 second pause. (Valid range = 1–60 seconds)

`play {`*`port`*`} {`*`filename`*`}`—Plays the specified voice file over the specified port. The port number is zero-based hexadecimal. For a phone connected to Port 9 on the PEU, type **`play 8 va\$va944.pcm`**. Because the Music on Hold Loop Tape is configured for continuous play, typing **`$va946.pcm`** generates a File Open! error message. When using the **`play {`*`port`*`} {`*`filename`*`}`** command, type **`r`** for repeat to hear the file.

`portscan`—Identical to sysinfo command.

`rec {1|2} {`*`filename`*`}`—Records a voice file of the file name specified using the Music1 or Music2 inputs on PEU A. After issuing the command, press the SPACEBAR to begin recording and ESC to end.

`relay {1|2} {`*`time`*`}`—Closes the Relay 1 or Relay 2 contacts on PEU A for the time specified. Note that the value is in hundredths of a second. For example, **`relay 1 100`** would close Relay1 for 1 second.

`ren {`*`filename`*`} {`*`newfilename`*`}`—Renames the specified file using the path and new file name specified. This command supports path names, but fails if the new file name already exists.

`rop {off|on|0|1}`—Controls the Ring Over Pager feature. With rop on, incoming CO calls ring over the Zone1 and Zone2 contacts on PEU A. Setting **`rop off`** disables this feature. The **Ring Over Pager** port property must be enabled in Port Configurations for this feature to work. (Default = OFF/0)

`set`—Displays many of the current system settings.

`set aaidletime={`*`n`*`}`—Sets the inactivity timeout for COs connected to Auto Attendants. If a DTMF tone is not detected within AAIDLETIME seconds, the VS1 system terminates the call on that CO. (Valid range = 60–3600 seconds, default = 600 seconds)

`set animode={0|1}`—Reserved for Telecor Technical Support use. Do not change this setting.

`set autopagetime={`*`n`*`}`—Sets the system default Auto Page time for ports configured to Auto Page. A call must ring at an extension for AUTOPAGETIME seconds prior to the first Auto Page. (Valid range = 0–120 seconds, default = 10 seconds)

`set autopagerepeat={`*`n`*`}`—Sets the system default Auto Page repeat interval. A setting of 0 disables Auto Page repeats. (Valid range = 0–120 seconds, default = 0 seconds)

`set aavolin={`*`n`*`}`—Sets the incoming DSP volume level for COs connected to voice channels (i.e. Auto Attendants, Voice Mail). This value is in addition to any change made through the dsp vol co ={n} in command. (Valid range = -12 to 12 dB, default = 0 dB)

`set aavolout={`*`n`*`}`—Sets the outgoing DSP volume level for COs connected to a voice channel (i.e. Auto Attendants, Voice Mail). This value is in addition to any change made in the dsp vol co ={n} out command. (Valid range = -12 to 12 dB, default = 0 dB)

`set cidformat={0|1}`—Sets the format of Caller ID in the SMDR files. Setting this value to 0 causes the area code to be enclosed within parentheses while a setting of 1 includes only the raw digits. (Default = 1)

`set coholdtime={`*`n`*`}`—Sets the system default held call and parked call recall timer. A held or parked call rings back to the station that placed it there after this timer expires. (Valid range = 20–3600 seconds, default = 300 seconds)

`set cpdans1={`*`n`*`}`—Reserved for Telecor Technical Support use. Do not change this setting.

`set cpdans2={`*`n`*`}`—Reserved for Telecor Technical Support use. Do not change this setting.

`set dialmindigits={`*`n`*`}`—Sets the minimum number of digits that must be dialed after pressing 9. Dialing fewer than DIALMINDIGITS results in the call being toll restricted. (Valid range = 1–11 digits, default = 4 digits)

`set dialtimerdigits={`*`n`*`}`—Sets the number of digits after pressing 9 at which the dial buffer switches from using DIALTIME1 to using DIALTIME2 as its timeout value. (Valid range = 1–11 digits, default = 7 digits)

`set dialtime1={`*`n`*`}`—Sets the number of seconds the dial buffer waits for digits after pressing **9** before assuming the dial string is complete. This value applies until DIALTIMERDIGITS have been dialed. (Valid range = 2–30 seconds, default = 20 seconds)

`set dialtime2={`*`n`*`}`—Sets the number of seconds the dial buffer waits for digits after pressing **9** before assuming the dial string is complete. This value applies after DIALTIMERDIGITS have been dialed. (Valid range = 2–30 seconds, default = 2 seconds)

`set dndtime={`*`n`*`}`—Sets the number of hours a standard phone or Modem/Fax port will remain on Do Not Disturb. The DND state is cancelled when this timer expires. Setting a value of **`0`** disables this timer. (Valid range = 0–999 hours, default = 1 hour)

set loop{*n*}={*file*}—This command changes the voice file source for the specified Loop Tape. For example, **set loop2=va\$va400.pcm** would change Loop2 to **$va400.pcm** at the end of current pass through Loop2. Loop Tapes should be started in the Loop Tapes section of the configuration program. (Valid range for {n} = 1–12)

set pcomwait={*n*}—Reserved for Telecor Technical Support use. Do not change this setting.

set ring0={*n*}—Sets the minimum duration for a valid CO power ring. This setting applies to the first ring only. Do not change this setting. (Valid range = 2–1000 hundredths of a second, default = 50 hundredths of a second)

set ring1={*n*}—Sets the minimum duration between valid CO power rings. Do not change this setting. (Valid range = 2–1000 hundredths of a second, default = 250 hundredths of a second)

set ring2={*n*}—Sets the maximum duration the system waits for the beginning of the next CO power ring. Do not change this setting. (Valid range = 2–1000 hundredths of a second, default = 500 hundredths of a second)

set smdrmode27={0|1}—This command sets the number of SMDR year digits. The default setting is **0**, which sets the number of year digits to four. A setting of **1** changes the year digits to two. It also removes the Field Description **Called Number** in Detail, Summary, and ACD SMDR files. A setting of **1** also removes the Field Description **Message Sent** from ACD SMDR files.

set zp{1|2}={loop1|loop2|audio1|audio2|silent}—Plays an audio source over the Zone1 or Zone2 paging contacts on PEU A. Specifying **loop1** or **loop2** plays the corresponding loop file, **audio1** or **audio2** plays the corresponding PEU music input, and **silent** turns the source off. Any page to Zone1 or Zone2 interrupts the specified audio source.

show acd {*acd number*}—Displays the status of the ACD number specified. (Valid range = 1–10)

show conn—Displays a list of existing connections on the VS1 phone system.

show events—Displays a list of Scheduled Events. The next event scheduled to run is preceded by a *.

show ports [*file*]—Displays a list of all devices configured on the system. Each line includes the port number, physical port type, software port type, extension number and port description. Optionally, a file name can be specified for output.

shutdown {*n*|idle}—This command shuts down the PBX and reboots the TVS in **{n}** minutes. Specifying the **idle** parameter resets the system as soon as no call activity is detected. (Valid range = 0–500 minutes, or IDLE)

stop—Identical to **shutdown 0** command.

stat—Displays a listing and status of system processes.

sysinfo—Displays the system software version, System ID (serial number), startup time, and other system information.

t1 comm {0|1}—Displays the status of the communications between the specified T1 Interface Card and the PBX. If one T1 Interface Card is installed, interface **0** corresponds to ports 97 through 128. If two T1 Interface Cards are installed, interface **0** corresponds to ports 65 through 96 and interface **1** corresponds to ports 97 through 128.

t1 error {0|1}—Displays a summary of the error information for the specified T1 Interface Card during the past 24 hours. Note: If one T1 Interface Card is installed, interface **0** corresponds to ports 97 through 128. If two T1 Interface Cards are installed, interface **0** corresponds to ports 65 through 96 and interface **1** corresponds to ports 97 through 128.

t1 setup {0|1}—Displays the configuration settings of the specified T1 Interface Card. Use the arrow keys to select a setting, the SPACEBAR to change a setting, and the ENTER key to save the changes. Press ESC to exit without saving the changes. If one T1 Interface Card is installed,

interface **0** corresponds to ports 97 through 128. If two T1 Interface Cards are installed, interface **0** corresponds to ports 65 through 96 and interface **1** corresponds to ports 97 through 128.

t1 stat {0|1}—Displays the current status of the specified T1 Interface Card. This information can be useful in T1 troubleshooting. If one T1 Interface Card is installed, interface **0** corresponds to ports 97 through 128. If two T1 Interface Cards are installed, interface **0** corresponds to ports 65 through 96 and interface **1** corresponds to ports 97 through 128. *See "T1 Interface Card Diagnostics" in Reference section for more information.*

time—Displays the current system time. See **date** command.

vec load {*file name*}—Reserved for Telecor Technical Support use. Do not change this setting.

vec dtmf {0|1}—Reserved for Telecor Technical Support use. Do not change this setting.

vm {*extension*}—Allows editing of the Voice Mail passcode and Pager Notification settings for the specified extension.

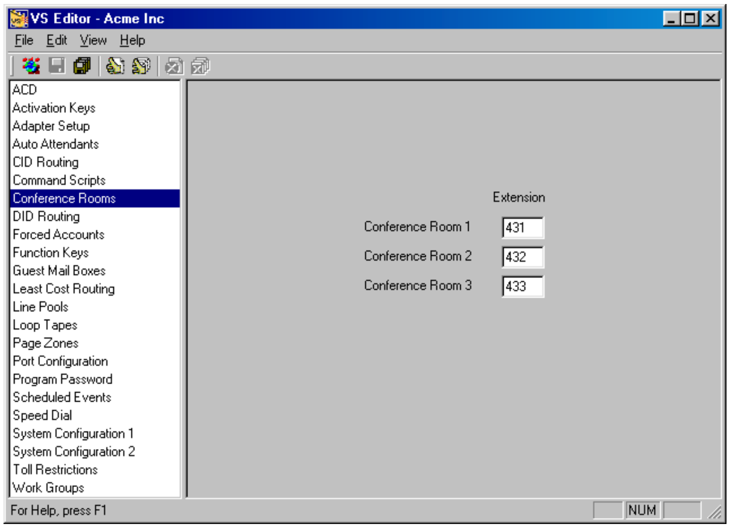
vmaint purge {*extension*|*} {*daysold*} [*forward to*]—Deletes all Voice Mail messages older then the number of days specified for the extension specified. Use the wildcard ***** in place of the extension to check every Voice Mail Box on the system. For example, **vmaint purge * 90** would delete every Voice Mail message on the system more than 90 days old. Optionally, the messages can be forwarded to Voice Mail by specifying a forward to extension.

vmaint scan [*forward to*]—Deletes Voice Mail messages that do not belong to any Voice Mail extension (such as orphaned messages). Optionally, the messages can be forwarded to another Voice Mail by specifying a forward to extension.

CONFERENCE ROOMS

RESET REQUIRED!

Conference Rooms on the VS1 System are the “meet me” type of conferencing. This means each person calls into the designated system Conference Room. The VS1 System has three 16-party Conference Rooms. These Conference Rooms can operate simultaneously, and can include any combination of inside or outside parties for a total of 16 participants.



Command Scripts pane

1. Select **Conference Rooms** in the Tree Control display.

   - The **Conference Rooms** pane appears.

By default, Conference Rooms are set up as extensions 431, 432, and 433. However, you can use any valid unused extension between 100–599 when using 3-digit extension numbers.

Calls can be transferred to a conference room by any VS1 Station. After transferring a caller to the Conference Room, the station user can dial the Conference Room extension to enter the conference. When a party enters an ongoing conference, a notification tone is played to alert attendees that someone new has joined the conference.

Conference Rooms can also be locked after a conference has been established so no other party can join. Any conference participant can lock a Conference Room by pressing **#1** on their phone, or unlock a conference room by pressing ***1**.

The VS1 System has two commands run from the TVS Command prompt that enable or disable Conference Room locking, and determine the length of time for each conference call.

`conf lock {0|1}`—Controls the ability to lock a 16-party conference room. A locked conference room results in a busy signal when its extension is dialed. When you type **`conf lock 0`**, Conference Room locking is disabled; **`conf lock 1`** enables Conference Room locking.

`conf timeout {`*`n`*`}`—Controls how long a conference call comprised of only CO ports can continue before the VS1 phone system terminates the call. Setting a value of 0 disables the Timeout timer. Pressing a DTMF key resets the timer. (Valid range = 0–32767 minutes, default = 15 minutes)

DID & DNIS ROUTING

Direct Inward Dialing (DID) allows a caller outside a company to call an internal extension without having to pass through an operator or an Auto Attendant. Dialed Number Identification Service (DNIS) works in a similar manner and is typically only available on inbound 800 and 900 service.

DID and DNIS on the VS1 System can only be used with a Telecor T1 Interface Card, and T1 service must be ordered from your local service provider. The dialed digits are passed down the line from the CO and the VS1 then completes the call to an extension. Your Service Provider will give you a block of DID or DNIS numbers to use, which are frequently provided in blocks of 100 numbers (e.g. 2001 to 2100). You must give your Service Provider the FCC ID number from the label on the T1 Interface Card. *For more information on T1, see "Installing a T1 Interface Card," in the Hardware section.*

In addition to DID and DNIS, the T1 Interface Card allows for Automatic Number Identification (ANI). ANI reports the number of the calling party. This is similar to Caller ID on an analog line. However, the only information provided is the number from which the caller is calling. Name information is not sent by the Service Provider.

The numbers received from the phone company can be a maximum of 15 digits for DID, DNIS and ANI. The VS1 system supports DID (or DNIS) and ANI with specific digit sequences sent by the Service Provider. Only when using both DID (or DNIS) and ANI do incoming signals need to be separated in order to be recognized by the VS1 system. For example, the digit sequences for the calling number (913) 888-7936 with the DID routing number of 274 are:

ANI and DNIS = * ANI * DNIS *	*9138887936*274*
ANI = * ANI **	*9138887936**
DNIS = ** DNIS *	**274*

Note If the Service Provider sends a pure digit stream (no * separating characters), the digits are considered ANI unless DID Routing is selected for the CO port.

Configuring a Port for T1 Service

There are three steps involved in configuring a port for T1 Service:

- Identifying the port for T1 Card usage (accomplished through Adapter Setup - see page 86.)
- Assigning the port as a CO Line.
- Configuring the port for DID or DNIS.

Assigning the Port as a CO Line

1. Select **Port Configuration** in the Tree Control display.
   - The **Port Configuration** pane appears.
2. From the **Configuration** drop-down box, select the configuration where you want to make changes.

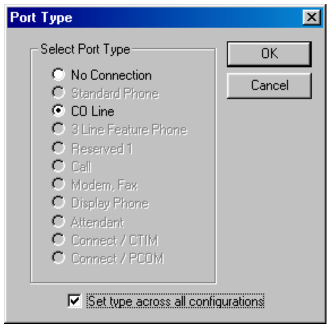
Note: If you have more than one configuration on your system you must set up each configuration.

3. Beginning at Port 97 (for one T1 Card) or Port 65 (for two T1 Cards) and ending at port 128, every fourth port is reserved, and the others are unassigned. The reserved ports are a function of

the T1 interface. Select the unassigned port that you want to assign as a CO port, and then click **Type**.

- The **Port Type** window appears.

4. Select **CO Line**, and then click **OK**.



Port Type window

5. Repeat steps 3 and 4 for each T1 port you want to assign as a CO Line.

6. To set up other configurations, repeat steps 2 through 7.

7. After all T1 ports are assigned as CO Lines, click the **Save** button in the toolbar to save your changes.

Configuring the Port for DID or DNIS

1. With a configuration selected in the **Port Configurations** pane, select a T1 port that you have assigned as a CO Line.

2. Enter the following information in the corresponding text boxes:

   Description: Enter a description for the T1 line.
   Extension: Enter an extension. (Enter **–1** if no extension is assigned)
   Number: Enter the digit **1** ten times, which function as placeholders.

Note A CO port configured for T1 service cannot be set up for both DNIS and DID.

3. Set up a default first ring target. This is important because if a call cannot be routed using the DID Routing table, it is routed based on the target rings and target port.

4. Under **Rings**, set the CO port to ring the first target for 10 rings.

   - Under **Target Ports**, set the CO port to ring the first Auto Attendant (AA1).

   - Check the **DID Routing** box.

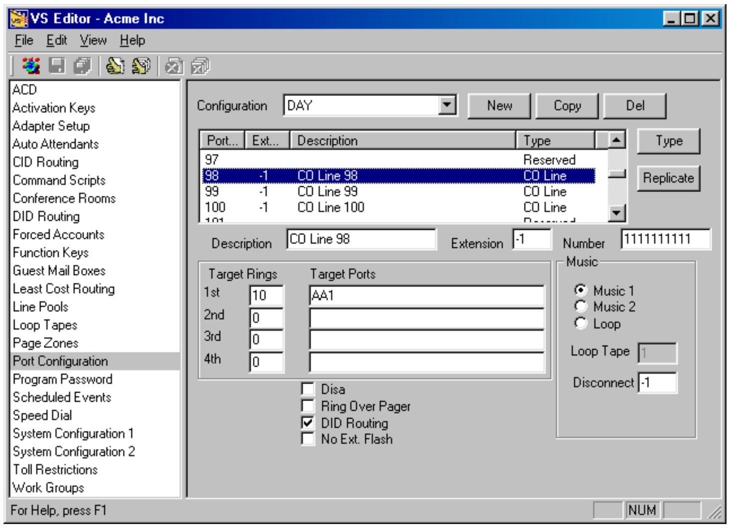
5. To set up additional ports with DID or DNIS routing, repeat the above steps *or* use the Replicate button and follow the on-screen instructions.

6. Set up additional configurations by following steps 1 through 5.

7. After all CO ports are configured, click the **Save** button in the toolbar to save your changes.

Note When using ANI only on the T1 port, **DID Routing** must remain unchecked.

Port configured for DID



Setting up a DID or DNIS Routing Table

To assign a block of numbers for DID routing, complete the following steps.

1. Select **DID Routing** in the Tree Control display.
   - The **DID Routing** pane appears.
2. Click **Add**.
3. In the **DID Entry** group box, enter the following information in the corresponding text boxes.

   Description: Enter a description for the block of numbers. The description is a 16-character message that is attached to the call. This message appears on all station equipment with a display.

   Start Num: This box must contain 7 digits. Enter the DID start number of your block of numbers preceded by the necessary amount of placeholders. Telecor recommends using 1 for the placeholder since it is then obvious that they are not part of the routing number.

   End Num: This box must contain 7 digits. Enter the DID end number of your block of numbers preceded by the necessary amount of placeholders. Telecor recommends using 1 for the placeholder since it is then obvious that they are not part of the routing number.

   Example (for a 4-digit DID): Start Number 1114001 **End Number** 1114100

   Start Time: Enter the start time that you want this block of numbers to recognized. Use standard 24-hour (military) time format.

   End Time: Enter the end time that you want this block of numbers to recognized. Use standard 24-hour (military) time format.

Note:	The **End Time** must not come before the **Start Time**.

Days: Select the days of the week that you want this block of numbers to be recognized.

4. Specify the **Target Ext** and **Target Mode**.

   Target ext: Enter the target extension for the DID Routing Entry.

   Fixed Target Mode: means that all numbers in the block are targeted to a specific extension. For example, assigning 1114001–1114100 to Extension 103 means that 4001 goes to Extension 103; 4002 goes to Extension 103, and so on.

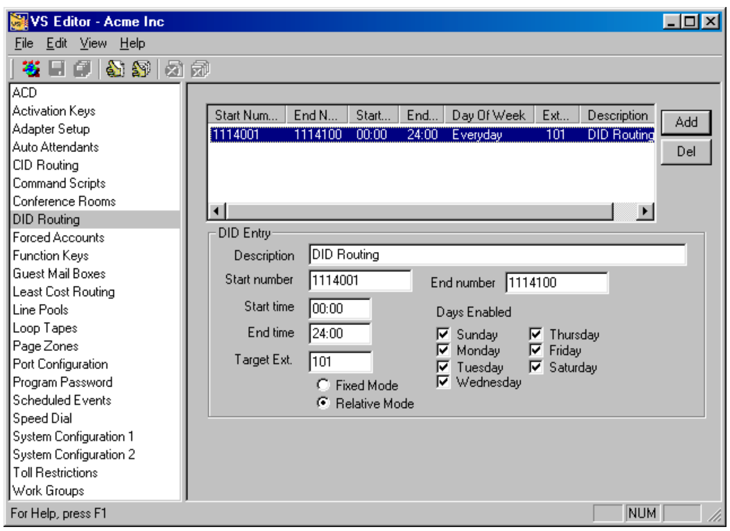
Note:	DNIS numbers are usually assigned as individual numbers and not in related blocks. DNIS numbers can only be routed in the Fixed Target Mode.

   Relative Target Mode: means that each number in the block is targeted to a specific extension. For example, if the Start Num is 1114001 and the Target Extension is 101, the number 4001 routes to extension 101, 4002 routes to extension 102, 4010 routes to Extension 110, and so on.

   - To assign one number to a specific extension, type the same number in the **Start Number** and **End Number** text boxes. In the **Target Ext** text box enter the target extension, and then select **Fixed** as the **Target Mode**.

5. Click the **Save** button in the toolbar.

6. The settings for the DID/DNIS Routing Table appear in the list box above. If there are any errors in your **DID Routing** entry, a dialog box appears notifying you of the error.



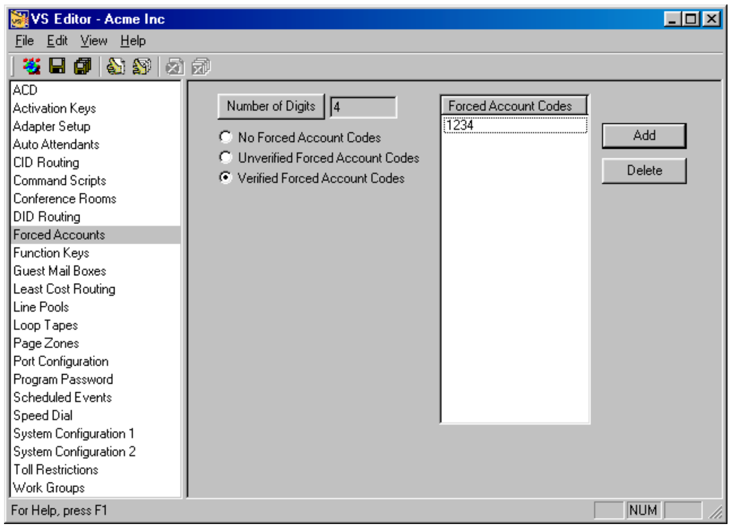
DID Routing pane

FORCED ACCOUNTS

RESET REQUIRED!

If changing the number of digits or changing between verified and non-verified

The Forced Account feature requires a phone user to enter a code to place an outgoing non-emergency call. After a number is entered the user hears four beeps to prompt input of an account code. If the required number of digits is *not* entered within 10 seconds, the call is terminated. If the required number of digits is entered, the code is compared with the defined list of Forced Account Codes. If there is no match, the user hears four beeps again and can re-enter the account code.



Forced Accounts pane

To create Forced Accounts, complete the following steps:

1. Select **Forced Accounts** in the Tree Control display.

   - The **Forced Accounts** pane appears. Three options are available:

2.1 Clicking **No Forced Account Codes** will prevent phone users from having to enter codes, even if their stations are enabled with Forced Account Codes in Port Configurations.

2.2 Clicking **Unverified Forced Account Codes** will force the phone user to enter the same number of digits specified in **Number of Digits** field – regardless of which digits – when dialing out. Click the **Number of Digits** button to enter the amount of digits (1 - 15) the user will be required to press.

2.3 Clicking **Verified Forced Account Codes** will force the phone user to enter a specific digit string when dialing out. Click **Add** to enter a digit string in the **Forced Account Codes** column. Multiple Verified Forced Account Codes can be created.

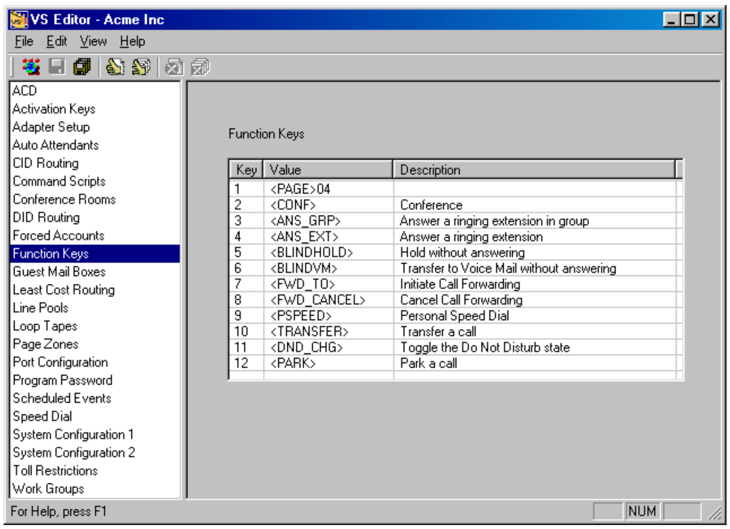
See "Setting up an Extension Port Type" on page 149 to enable Forced Account Codes on an extension.

FUNCTION KEYS

RESET REQUIRED!

Function Keys enable DP200 Phone users to access VS1 System features (such as Call Forwarding, Paging, Transfer, and Do Not Disturb) via the DP200 Feature Buttons. There are 12 Function Keys available, and the default functions of these can be programmed with the VS1 Editor.

Note Telecor uses the terms Function Keys and Feature Buttons interchangeably. The term Function Key is used primarily when programming with the VS1 Editor. The term Feature Button refers to one of the 12 corresponding buttons on the DP200 Phone.



Function Keys pane

The 12 Function Keys are programmed by default as follows:

F1=Page All	F7=Forward To
F2=Conference	F8=Forward Cancel
F3=Pickup Group	F9=Personal Speed Dial
F4=Pickup Extension	F10=Transfer
F5=Blind Hold	F11=Do Not Disturb Toggle ON/OFF
F6=Blind Transfer to Voice Mail	F12=Park

To change a function of a Function Key, complete the following steps:

1. Select **Function Keys** in the Tree Control display.

   - The **Function Keys** pane appears with a list box that shows a list of 12 Function Keys that correspond with the 12 Feature Buttons on a DP200.

2. Select a function under the **Value** column and a drop-down box appears

   or

2. Select a description under the **Description** column and a drop-down box appears.

3. Select a new function for the Function Key.

   - If a new value was selected, the Description changes.

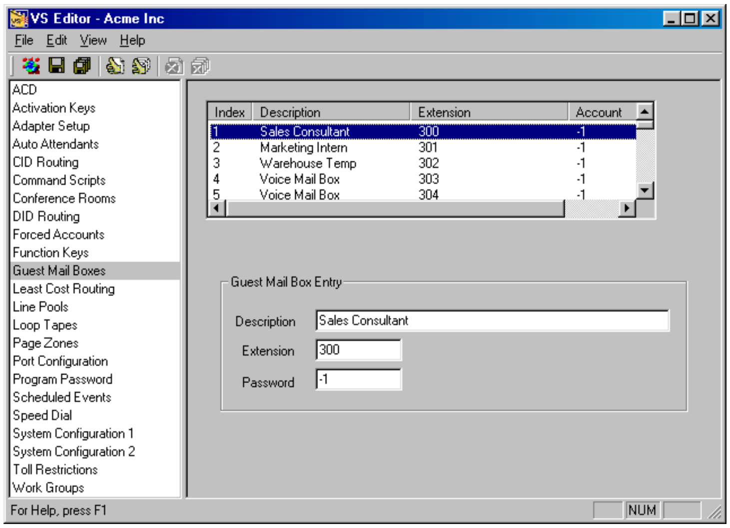
- If a new description was selected, the Value changes.

Note	If a DP200 user has customized a Feature Button (*see the DP200 Quick Reference Guide*) to perform a different function than what the default is set to, changing Function Keys with the VS1 Editor has no effect.

GUEST MAIL BOXES

RESET REQUIRED!

The VS1 Telephone System is configured with the possibility of a maximum of 200 Guest Mail Boxes to store Voice Mail messages. A Guest Mail Box is a virtual port for individuals without a station option connected to the system. Guest Mail Boxes are assigned an extension number just as extensions with devices and function the same as Voice Mail for extensions connected to the system. You can access Guest Mail Boxes from any internal station option, or from an outside line.



Guest Mail Boxes pane

To create a Guest Mail Box, complete the following steps:

1. Select **Guest Mail Boxes** in the Tree Control display.

   - The **Guest Mail Boxes** pane appears. The first 20 Guest Mail Boxes in the list box are set to 300 to 319 by default. The remaining Guest Mail Boxes are set to –1, indicating that these are inactive. The default Guest Mail Boxes can be modified if required.

2. Select an inactive Mail Box in the list box

3. In the **Guest Mail Box Entry** group box, enter the following in the corresponding text boxes:

   Desc: Enter a description of the Guest Mail Box. This description appears on display phones or CTI station options.

   Ext: Enter an unreserved extension for the Guest Mail Box. By default, extensions 300 to 319 are reserved for Guest Mail Boxes. *See "Default Extensions and Access Codes" in Reference section.*

   Password: If required, enter a DISA password for this Guest Mail Box. The DISA password must be a 5-digit number to work correctly.

Note	Make sure DISA is enabled on at least one CO port to use this feature. *See "Direct Inward System Access" in the Reference section for more information.*

4. Click the **Save** button in the toolbar.

   - The Guest Mail Box entry is added to the list box above.

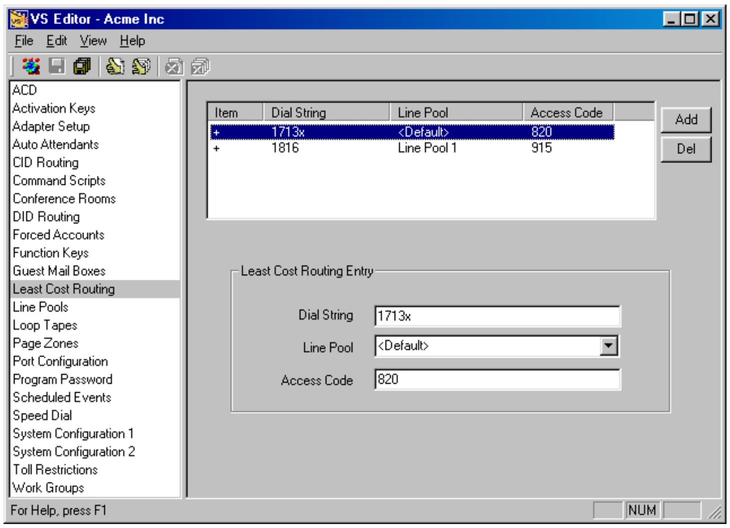
5. Repeat the above steps for other Guest Mail Box entries.

LEAST COST ROUTING

Least Cost Routing (LCR) automatically selects a low-cost phone line based on the number dialed. Least Cost Routing (LCR) looks at the dial string that a person dials from their station, prefixes the number with the long distance carrier access code (if defined), and then routes that call to a specified Line Pool.

Least Cost Routing on the VS1 System does not guarantee the least expensive method of call routing. Cost savings are incurred when low rate long distance carriers are listed in the **Least Cost Routing** window. As long distance carrier rates fluctuate, the **Least Cost Routing** window must be maintained to reflect the current low-cost carrier for a given area code.

Least Cost Routing pane



To create an entry for Least Cost Routing, complete the following steps:

1. Select **Least Cost Routing** in the Tree Control display.

   - The **Least Cost Routing** pane appears.

2. Click **ADD**.

3. In the **Least Cost Routing Entry** box, enter the following in the corresponding boxes:

   Dial String: Type the digit string you want the system to match. The Dial String can be an entire phone number or any prefix set of identifying numbers such as an area code. Adding an **x** represents a single digit, matching any number 0-9. For example, 1713x or 1816x.

   Line Pool: Select a Line Pool that the call will go out on.

   Access Code: Type the long distance carrier access code, which will be prefixed to the dialed number.

4. Click the **Save** button in the toolbar.

   - The LCR entry is added to the list box above.

5. Repeat the above steps for other LCR entries.

Least Cost Routing is enabled or disabled in each individual Extension port type configuration. *See "Setting up an Extension Port Type" on page 149 for more information.*

Note	The default number of Least Cost Routing entries is 1200. This number can be adjusted in the System configuration 1 pane.

How a Long Distance Call is Routed

The LCR feature on the VS1 System performs a "starts with" search for outbound calls, which means LCR does a First Match, Top Down search through Dial Strings. The **Dial String** sorts by 0 through 9, and then the wildcard **x**. Make the most economical or preferred entries first and in descending order, as the list moves from Top Down. It is the installer's responsibility to determine carrier rate values.

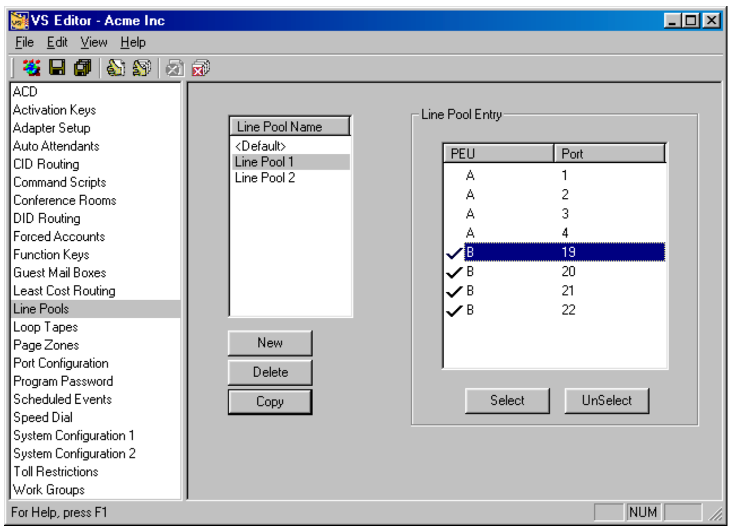
The LCR checking process is shown in the following example:

1. The station user dials a phone number.
   - A check of the extension's toll restrictions is performed. If not toll restricted then…
2. If LCR is enabled, a check of the **Least Cost Routing** window is performed from the Top Down.
3. When the first match is found in the window, any digits entered for the Access Code are prefixed to the Dial String and then the call is routed through the LCR Line Pool.
4. If a match is not found in the window, the call is routed through the station user's Line Pool and Access Code.

LINE POOLS

Line Pools are used in PBX phone systems to assign a group of CO lines to a station option. With the VS1 System, you can assign up to 96 COs to a specific Line Pool. Generally, PBX phone systems have a higher ratio of phones than actual CO ports. Pooling station users together for certain ports gives them more equal access to those phone lines. If all the available CO ports are in use, the VS1 System generates a busy signal, informing the caller that all the lines are in use at the time.

CO ports in a Line Pool are always arranged in reverse order (descending numbers). It is important to remember this when assigning specific CO lines to ports. No matter what order you select CO ports, the system always arranges outgoing calls to go out on Line Pools in reverse numerical order. For example, if you have four CO lines, and you select all four for your Line Pool, the system sets up the calls to go on CO line 4 first, then 3, 2, 1. Generally, incoming calls start at the top of the CO ports and move down. Reverse numerical order helps reduce call crashing, or connecting to a call without it ringing a station first.



Line Pool pane

To add CO Lines to a Line Pool, complete the following steps:

1. Click **Line Pools** in the Tree Control display.
   - The **Line Pools** pane appears.
2. The **Line Pool Name** list box consists of the **<Default>** Line Pool. The **<Default>** Line Pool is a permanent Line Pool, which is automatically placed in an Extension port type configuration. System installers can use or modify the **<Default>** Line Pool or create new Line Pools according to site configurations.
3. Click the Line Pool you want to modify or, to create a new Line Pool, click **New**.
   - The **Line Pool Entry** list box shows all the available CO ports you can use for Line Pools.
4. Click on a CO port you want included in the selected Line Pool and click **Select**.
   - A check mark indicates that the CO port is selected. A CO port must have a check mark in order to be included in a Line Pool. A CO port can be in some, none or all Line Pools.

5. Click the **Save** button in the toolbar.

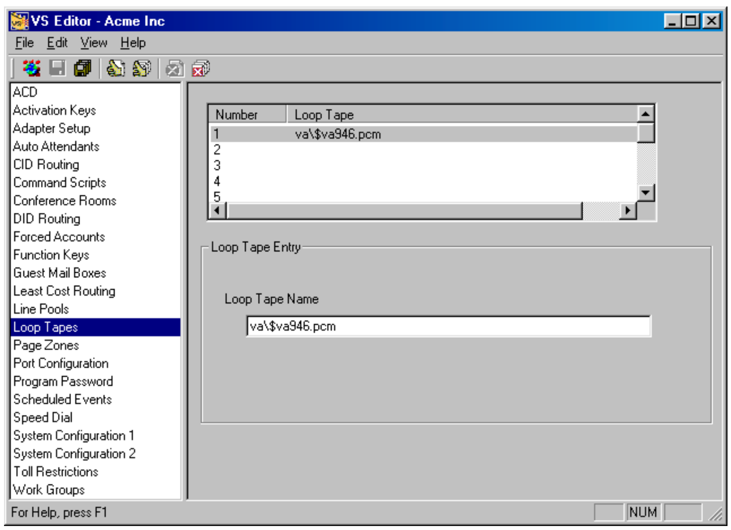
Line Pools are put into action in each individual Extension port type configuration. *See "Setting up an Extension Port Type" on page 149 for more information.*

LOOP TAPES

RESET REQUIRED!

A Loop Tape consists of a voice or sound file that an outside caller hears while waiting in an ACD queue or while on hold at an extension. From one to 12 Loop Tape messages can be created for use at any given time. Each Loop Tape, whether in use or not, permanently plays during system operation and uses the resources of one voice channel. Because there can be a maximum of 12 voice channels there can only be a maximum of 12 Loop Tapes.

Warning!	Voice channels are used to read and write voice files and are used every time an Auto Attendant or Voice Mail is accessed. The VS1 System comes with a System Activaiton Key for four voice channels. Additional voice channels are available in increments of four, up to a maximum of 12. Ensure that there are always less Loop Tapes than Voice Channels on the System.



Loop Tapes pane

To create a Loop Tape, complete the following steps:

1. Select **Loop Tapes** in the Tree Control Display.
   - The **Loop Tape** pane appears.
2. Select a Loop Tape Number in the **Loop Tape** list box. By default, Loop Tape 1 is set to play file **$va946.pcm**. This file was created for systems without a music source input. It plays a generic on-hold music file and an on-hold courtesy message.
3. A voice file needs to be assigned to the Loop Tape number. As an example, assume the voice file is 500. In the **Loop Tape Name** text box, enter the voice file in the following way:

   va\$va500.pcm

4. Click the **Save** button in the toolbar.

Note	Any files entered in the **Loop Tape** pane permanently use the resources of one voice channel on the system.

To record a voice file, see “Recording Voice Files” in the Reference section.

To apply a Loop Tape to an ACD, see "Configuring an ACD" on page 90.

To apply a Loop Tape to a CO Line port, see "Setting up a CO Port Type" on page 147.

Changing a Voice File Playing as a Loop Tape

While a voice file is playing as a Loop Tape you can listen to it but you can't record over it to change the message. To change the recorded message, you must record a voice file with an unused number and copy it into the voice file number playing as a Loop Tape. This is accomplished using the TVS Command Prompt via the Terminal Window of Tel-Site.

Complete the following steps, assuming that voice file **`$va500.pcm`** is playing for Loop Tape 1.

1. Record a new voice file with the file number **`$va501.pcm`**. *(See "Recording Voice Files" in the Reference section).*

2. At the TVS Command Prompt, type **`set loop1=va\$va501.pcm`**, and then press ENTER.

   - At this point you must wait for the original Loop Tape 500 to stop running. If that Loop Tape is one minute long, then you must wait at least that long before you proceed. At that point the voice file for Loop Tape 1 is changed from 500 to 501.

3. Type **`copy va\$va501.pcm va\$va500.pcm`** , and then press ENTER.

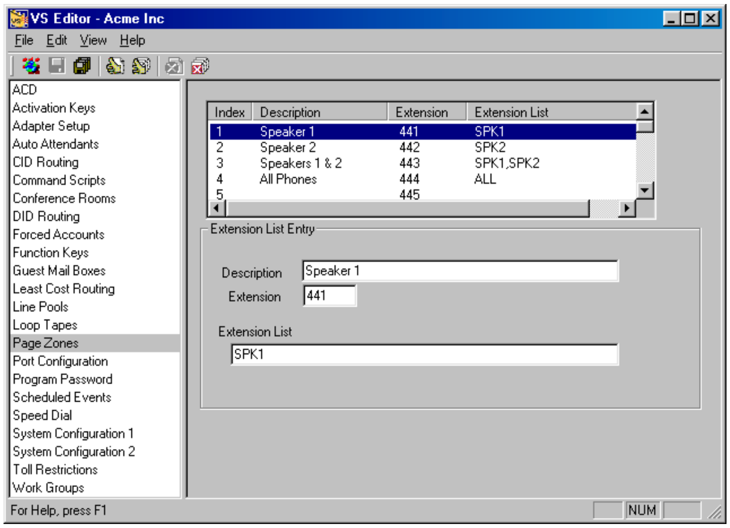
   - This copies the new voice file 501 into the original voice file 500. No system reset is required for this operation.

4. Then type **`set loop1=va\$va500.pcm`** to set the voice file number back to the original voice file number 500.

PAGE ZONES

RESET REQUIRED!

The VS1 System has 20 software Page Zones that can consist of any combination of DP200 speakerphones and the two overhead paging systems connected to a PEU 205 or DCU. Each Page Zone is assigned a separate extension.



Page Zones pane

To create a Page Zone, complete the following steps:

1. Select **Page Zones** in the Tree Control Display.

2. The **Page Zone** pane appears. By default, the VS1 System allocates extensions 441 to 460 for Paging Zones. In addition, four Paging Zones already exist:

   - **Speaker 1** is extension 441 and consists of the overhead paging system connected to the first zone output of a PEU 205 or DCU.
   - **Speaker 1** is extension 442 and consists of the overhead paging system connected to the second zone output of a PEU 205 or DCU.
   - **Speakers 1 & 2** is extension 443 and consists of both overhead paging systems connected to the zone outputs of a PEU 205 or DCU.
   - **All** is extension 444 and consists of all DP200 speakerphones and the two overhead paging systems connected to the zone outputs of a PEU 205 or DCU.

3. To modify an existing Page Zone select it from the list box. To add a new Page Zone, click the empty space in the list box.

4. In the **Extension List Entry** group box, enter the following in the corresponding text boxes:

   Description: Type the Page Zone description, e.g. **Sales**.

   Extension: Type the extension of the Page Zone.

   Extension List: Type the extensions you want included in the Page Zone. Up to 19 extensions can be listed in this text box. Use commas to separate the extensions. Do not use spaces between the extensions. For example, **101,102,103,104,105**. The extension names for the two hardware connections on the PEU (or DCU) are **spk1** and **spk2** respectively. Type **ALL** to include all DP200 speakerphones and both overhead paging systems.

5. Click the **Save** button in the toolbar.

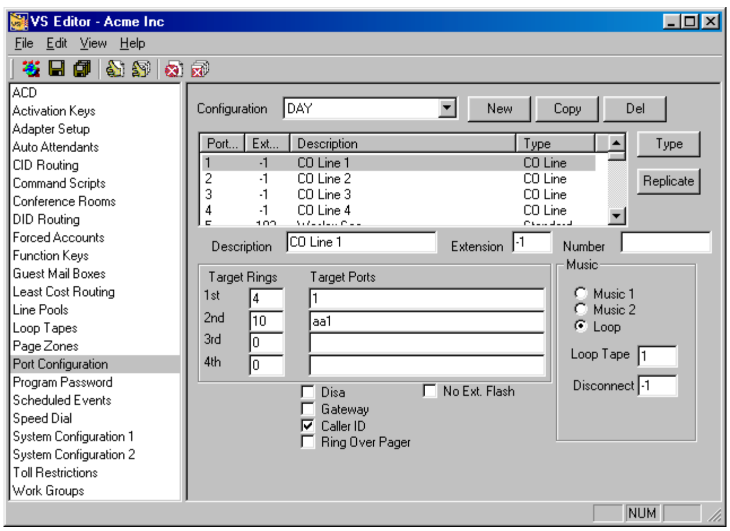
6. Repeat steps 3 and 4 for other Page Zone entries

PORT CONFIGURATIONS

A Port Configuration allows CO Lines and Extensions to be set up for functioning on the VS1 System. Many other features of the VS1 Editor, such as such as Toll Restrictions, Line Pools, and Least Cost Routing, are put into action through Port Configurations. A Port Configuration ties other VS1 Editor panes and settings together to create an operational Configuration.

By default, the VS1 Telephone System only requires one Configuration to operate—the Day Configuration. The Telecor Voice Server (TVS) loads the Day Configuration automatically every time it starts—regardless of Scheduled Events or any other system features. The Day Configuration determines which port type runs during business hours for each port. However, you are not limited to only one Configuration. The VS1 System supports as many Configurations as you can create, limited only by hard disk space. Configurations can be created for Night, Weekends, Break, and so on.

Note	A Day Configuration must be created for each new site. Without a Day configuration, the TVS will not function properly.



Port Configuration pane

Creating a New Configuration

1. Click **Port Configurations** in Tree Control Display.

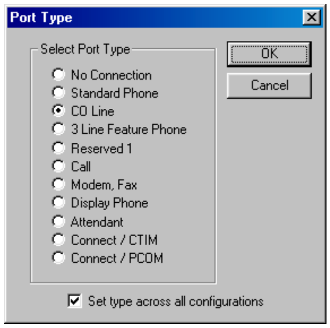
   - The **Port Configuration** pane appears.

2. Click **New**.

   - The **Configuration Name** dialog box appears.

3. In the **Enter Configuration Name** text box, enter a name or description (e.g. Day) for the configuration and click **OK**.

4. Assign each port a type by selecting the port you want, and then clicking **Type**.



Port Type dialog box

RESET REQUIRED!

- The **Port Type** window has the following options.

  No Connection
  Standard Phone
  CO Line
  3 Line Feature Phone
  Reserved 1
  Call
  Modem/Fax
  Display Phone
  Attendant
  Connect/CTIM
  Connect/PCOM

For example, assume there is a 16-port system with one Port Expansion Unit (PEU) installed. If the jumpers on the PEU are set so the first 4 ports are COs and the last 12 ports are Extensions, set your port types accordingly.

When assigning a type to a port, the type can be set across all configurations by checking the **Set type across all configurations.**

Note In the Port Type window, **Call** and **3 Line Feature Phone** are reserved for backwards compatibility.

Note To save time, multiple ports can be selected at once and assigned the same type. Holding down the SHIFT key allows consecutive ports to be selected. Holding down the CTRL key allows non-consecutive ports to be selected.

Copying a Complete Configuration

To save time when configuring a system installation, you can copy a current working configuration into a new configuration, and then make your changes accordingly. To copy a current configuration into a new configuration perform the following steps:

1. In the **Port Configurations** pane, click **New**.

   - The **Configuration Name** dialog box appears.

2. In the **Enter Configuration Name** text box, enter a name or description for the configuration and click **OK**.

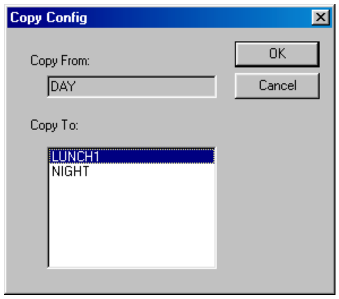
- At this point, the new Port Configuration appears with all the ports unassigned.

3. In the **Configuration** drop-down box, select the Configuration whose information will be copied.

4. Click **Copy**.

    - The **Copy Config** dialog box appears.

Copy Configuration dialog box



5. In the **Copy To:** box select the newly-created Configuration that information will be copied to.

6. Click **OK**.

7. The **Confirm** window appears asking if you want to copy over the existing information in your unassigned configuration. Click **Yes**.

Replicating Individual Port Settings

To save time when configuring individual ports, all selections from a configured port can be replicated to a port of the same type – in the same configuration or other configurations -- leaving you with only minimal changes to the new port. Complete the following steps:

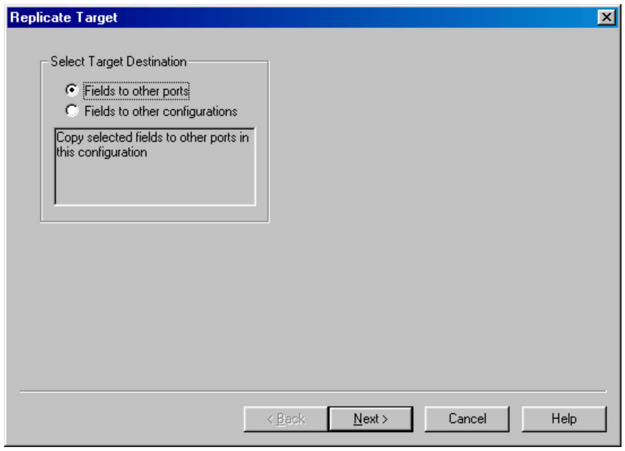
1. Select a configured port.

2. Click **Replicate**.

    - The **Replicate Target** dialog box appears. Two options are available:

      Field to other ports: Copies selected field to same port types in the configuration.

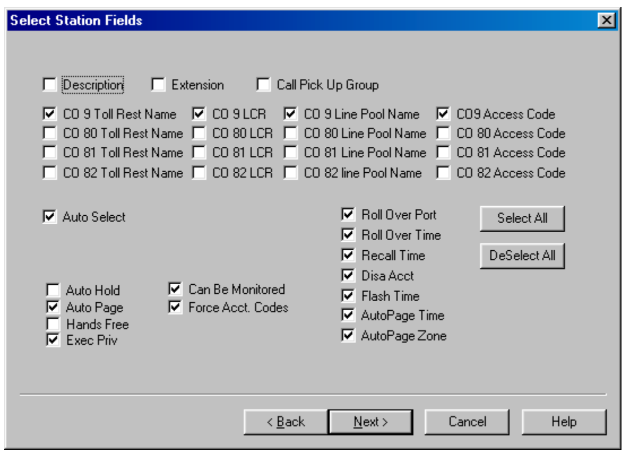
      Fields to other configurations: Copies selected field to the same port number in other configurations. The targeted port number **must be set** to the same port type as the source.



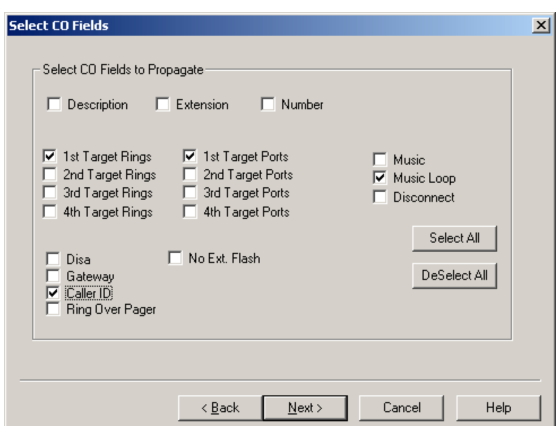
Replicate Target dialog box

3. Select the appropriate option and click **Next**.

   - The **Select Station Fields** or **Select CO Fields** dialog box appears.



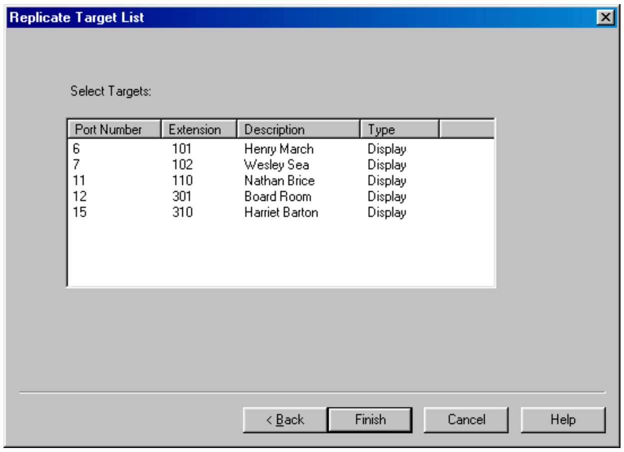
Select Station Fields dialog box



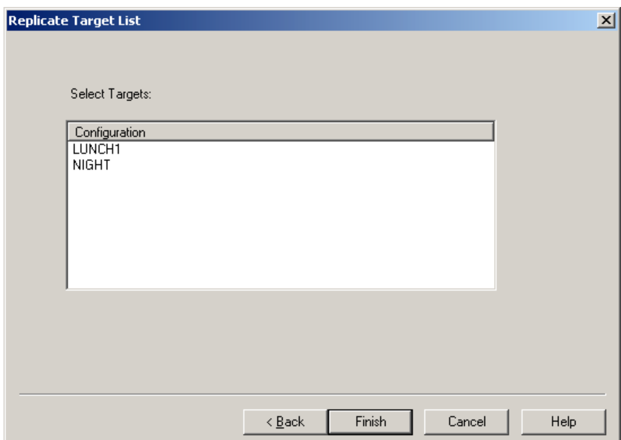
Select CO Fields dialog box

4. Check the options that are to be copied and click **Next**.

- The **Replicate Target List** dialog box appears.



Replicate Target List “Fields to other Ports”

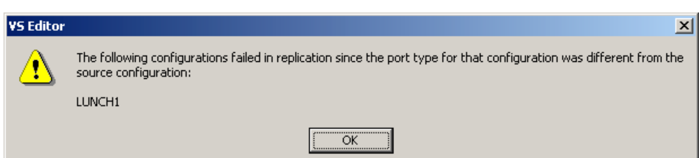


Replicate Target List “Fields to other Configurations”

5. Select the target ports (if **Fields to other ports** was chosen) or the configuration (if **Fields to other configurations** was chosen) that are to receive the replicated information.

6. Click **Finish**.

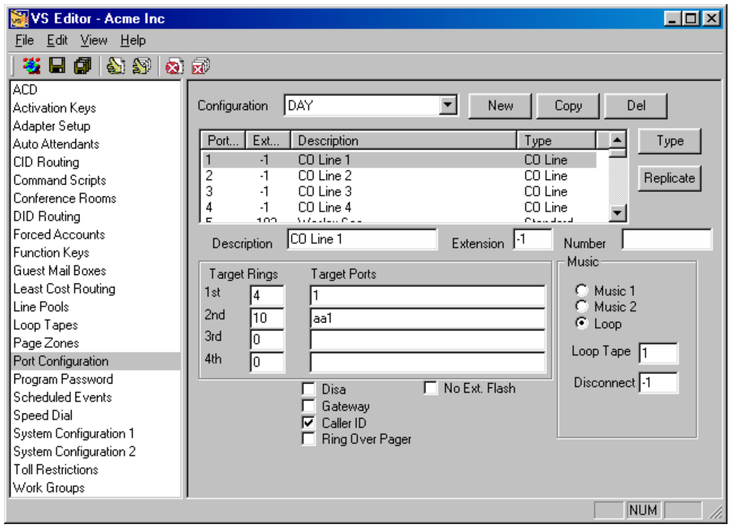
Note	For replicating Fields to other configurations, if the port type of the destination does not match the port type of source, a warning appears stating that replication failed.



Replication failure if destination does not match source

Setting up a CO Port Type

Setting up a CO port involves filling in the accompanying text and check boxes. What follows is a description of the boxes and how that information is used by the VS1 System. Click the **Save** button in the toolbar when information has been changed or entered.



CO Line

Description: Explains what the CO port is used for. When a DP200 display phone or CTI station option user gains access to the CO line, the user sees the characters entered in this text box.

Extension: The default extension for the CO Port is set to **–1**. Telecor recommends that this not be changed. Doing so helps prevent toll fraud. If an extension is assigned to the CO Port, then anyone can dial that CO Port extension from any extension on the system, or through an Auto Attendant. They can then directly gain access to dial tone and bypass any toll restrictions.

Note If your site needs the capability to perform PSTN Call Forwarding, you must assign an extension to the first CO port in the hunt group. Then dial that CO port extension to get dial tone and configure the PSTN Call Forwarding.

Number: This text box is generally used to enter the telephone number for the CO port. That number is used for two purposes: by the Attendant station users to pop **Greetings** windows based on the number called and for DID Routing.

Target Rings and **Target Ports** are used in conjunction with each other when a call rings a CO port. At the **1st** Target the calls ring the Ports listed under **Target Ports** for the number of times specified under **Rings**. If the call is not answered in the specified number of rings, the call drops to the **2nd** Target, where it rings the Ports listed under its **Target Ports** for the specified number of rings.

A total of 20 Ports can be entered in one **Target Ports** text box. The Ports must be separated by commas and without spaces (e.g., 7,8,9,10). All the extensions configured for the Ports listed in the Target Ports text box ring simultaneously. For example, if Ports 7,8,9,10 are listed under the **1st Target Ports**, the extensions configured for ports 7, 8, 9, and 10 all ring at the same time. Calls that ring through the **4th Target Ports** without being answered are moved back up to the **1st Target Ports** to try again.

You cannot use a Port configured for a CO Line in the Target Port text box. The following are also valid entries:

AA1 to **AA20**: (Auto Attendants 1–20)
ACD1 to **ACD10**: (ACDs 1–10)
Null: Null rings nowhere. The null device is useful for configuring DISA.
VM{n}—VM{n}: would send the call to the Voice Mail of Extension **{n}**. Any valid Voice Mail extension can be used after VM.

Properties Group Box

DISA: Enables Direct Inward System Access to internal dial tone from an external telephone not connected to the VS1 Phone System. Many VS1 System installers choose to make DISA available on only one CO port so that it is not generally accessible to every end-user at a particular site. Providing access to a limited number of people can help reduce fraud and abuse of the DISA feature. In addition the CO Port must be targeted correctly *(see "Direct Inward System Access" in Reference section*) and an Extension must be configured with a DISA Account (*see "Setting up an Extension Port Type" on page 149*).

Note It is more effective to use a CO port that is not in a hunt group so that the CO number is always available to the caller wanting DISA access. If the CO port is part of a hunt group, it should be in the middle of the hunt group.

Gateway: This is used in conjunction with the **Disconnect Signal** property of an Extension port. Enabling this option causes the CO port to respond to a "D" DTMF tone as disconnect. It is not ordinarily used with a CO port. When selected, and the system hears a "D" DTMF tone, it automatically disconnects the call.

Caller ID: When selected, this option enables the system to add additional ring pauses before it starts ringing the Target Ports. The additional ring pauses allow the Target Ports time to wait for the Caller ID.

Ring Over Pager: If a call rings the CO port, an audible ring is heard over a paging system connected to the first Zone Pager Output of a PEU-205. The call can then be picked up from any extension by dialing **#8** and the extension of where the call is ringing.

Music 1: When a caller on the CO Line is placed on hold, the music source connected to Music Input 1 on PEU 205 or DCU is played for the caller. If an internal call is placed on hold the caller will always hear the Music 1 source.

Music 2: When a caller on the CO Line is placed on hold, the music source connected to Music Input 2 on PEU 205 or DCU is played for the caller.

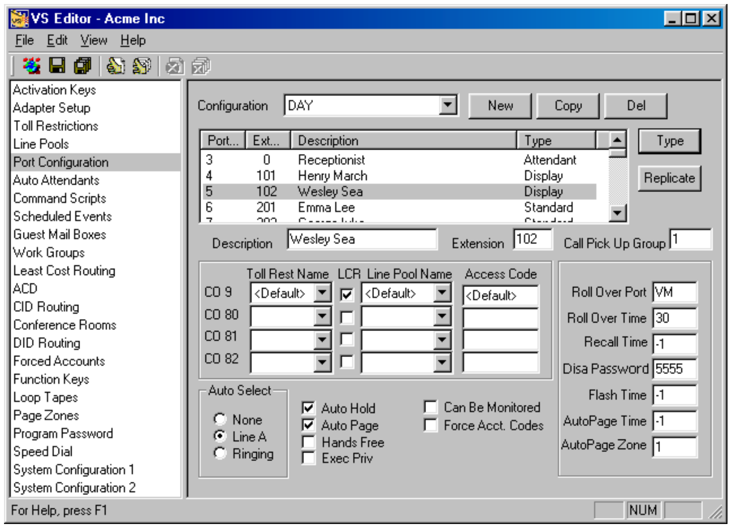
Loop Tape:When a caller on the CO Line is placed on hold, the Loop Tape number entered in the **Loop Tape** text box is played for the caller.

Disconnect: The default setting is **–1**, which keeps the default setting for the **Default Disconnect (.01) sec** text box in System Configuration 1. The Default Disconnect in System Configuration 1 is a global setting and is set to .04 sec, or 400 milliseconds. If you change the CO port **Disconnect** text box, it only affects the specific CO port. Any drop in loop current equal to or larger than the defined time results in a disconnect.

No Ext Flash: Disables external hookflashes on the CO port. Do not enable this feature on CO ports that require external hookflashes, such as Centrex.

Setting up an Extension Port Type

Setting up a Extension port involves filling in the accompanying text and check boxes. What follows is a description of the boxes and how that information is used by the VS1 System. Click the **Save** button in the toolbar when information has been changed or entered.



Extension port (DP200)

Description: Enter a name or description for the Extension Port. When a DP200 display phone or CTI station option user gains access to the extension, the user sees the characters entered in this text box.

Extension: Enter an extension for the port. If an extension is assigned to the port, then anyone from any extension on the system, or through an Auto Attendant, can dial that extension.

Call Pick Up Group: This option is used in conjunction with the feature keystroke of ***8**. The ***8** feature keystroke allows a user to pick up a ringing extension in a group. A group consists of multiple extension ports with identical numbers entered in the **Call Pick Up Group** text box. This box is set to **1** by default, but the number can be from **0** to **9**. Entering **–1** means this station option does not use groups.

If the site environment has more than one department housed in the same area, a group can be assigned to each department so that only those people in the same group/department have the option of picking up calls. For example, the Sales department is set to 1 and the Service department is set to 2. When a call is ringing a Sales phone, another user in the Sales department can dial ***8** from a Sales phone to answer the call. If a user is on a Sales phone and hears a Service phone ringing, the user is not able to pick up that Service call by dialing ***8**.

Note You can set up a Command Script (GRPID OFF/ON) to enable or disable Call Pick Up Group during a specific time. For example, you could run a Command Script to turn on the Answer Group function when the system runs the Day configuration.

Dial Groups

A Dial Group determines how an external call made from an extension is routed. A Dial Group includes the following boxes: Toll Rest Name (Toll Restriction Name), LCR (Least Cost Routing), Line Pool Name, and Access Code #. The VS1 Phone System has four Dial Groups available for extensions: **CO 9**, **CO 80**, **CO 81**, and **CO 82**. An extension user accesses a CO Line to make an external call by pressing the respective Dial Group number and then dialing the phone number. The call is then routed based on the settings for the Dial Group. The most common Dial Group used is CO 9. By default, CO 9 is set up for use on the VS1 System, and is described below.

Toll Rest Name: Select a Toll Restriction from the drop-down box. The default Toll Rest Name is **<Default>**.

LCR: Uses the Least Cost Routing pane for routing calls. Checked by default.

Line Pool Name: Select a valid Line Pool for the Dial Group. The default Line Pool Name is **<Default>**.

Access Code #: If **<Default>** is placed in this text box, it tells the system to use the **Default CO Access Code** text box entry found in the **System Config 1** pane. Generally, the Access Code # is used for Centrex services where you have to dial 9 for every outbound call. Putting a 9 in the **Default CO Access Code** text box in the **System Config 1** window automatically prefixes outbound calls with a 9 every time.

RO Port: Stands for Rollover Port. Used by an extension when a call goes unanswered. By default, the RO Port is set to **VM**, which means unanswered calls go to the port's Voice Mail. Other valid entries for this text box include the following: any other station port, **VM{ext}** for an extension's Voice Mail (with **{ext}** representing actual extension), **AA1** to **AA20** for Auto Attendants 1 to 20, and **ACD1** to **ACD10** for Automatic Call Distribution 1 to 10. **0** is also a valid RO Port, which sends a call back to the station that transferred the call.

Note	Extension numbers, including Guest Mail Boxes and Work Groups, are not valid Rollover Ports.

RO Time: Stands for the Rollover Time, which is the time in seconds an external call ringing at a station rings before it is transferred to the RO Port. The default is set to 30 seconds, or approximately five rings. The minimum value that can be entered is 1 second and the maximum is 999 seconds.

Recall Time: By default, the Recall Time is set to **–1**, or 300 seconds. Recall Time sets the time in seconds a call you place on Hold or Park remains on Hold or Park before ringing back to your station. To change the default, see the CO Hold Time command in Command Scripts. Recall Time is not supported by CTI Attendant and Connect station options. Recall Time is not applicable for extension to extension calls.

DISA Password: DISA Password is a five-digit number ranging from 00000–99999. The **DISA Password** text box is where an account number is specified for the extension to validate DISA operation. By default, the DISA Acct is set to **–1**, which means no account has been assigned, and no DISA features are available for that extension.

Flash Time: Flash Time sets the maximum threshold for a hookflash on an extension. By default, Flash Time is set to 0 for DP200 Display Phones, which means it ignores analog hookflashes. For all other port types, Flash Time is set to **–1**, which by default is 750 milliseconds. Flash Time must be greater than 500 milliseconds. A valid hookflash is considered to be greater than 500

milliseconds and less than the Flash Time. Flash Time is entered in one hundredths of a second (80 = 800 milliseconds).

AutoPage Time: AutoPage Time tells the system how long to wait from the time a call begins ringing at an extension before it issues its first AutoPage. By default, AutoPage Time is set to **–1**, which is 10 seconds. After 10 seconds expire, a page is made announcing "Call waiting" and the extension where the call is ringing. You can set AutoPage Time from 1–999 seconds.

Note	Make sure the AutoPage Time is set to a shorter amount of time than the RO Time. Otherwise, calls automatically rollover to the RO Port before the message is played.

AutoPage Zone: The AutoPage Zone is the Page Zone index number where automatic paging occurs. By default, AutoPage Zone is set to **1**. You can set AutoPage Zone from 1 through 20.

RESET REQUIRED!

Auto-Select

Auto Select determines which line is selected on multi-line phones such as the DP200 Display Phone and the Connect CTI client application. Auto Select does not affect single line phone operation. By default, the **Line A** option is checked. Three options are available:

None – phone user must pickup the receiver and press the 'line' button he wishes to use Line 1 through 5

Line A – the line is selected automatically when the handset is lifted or the speakerphone button is pressed.

Ringing – the ringing line is automatically selected when the handset is lifted or the speakerphone button is pressed.

Note	The **Line A** option must be selected for Telecor Connect using PCOM port types for the phone to function when the computer is turned off.

Properties

Auto Hold (only for DP200 phones): Automatically places a call on hold when a DP200 display phone user presses another line appearance button to answer a call. If Auto Hold is disabled, the first line appearance call is disconnected when the new call is connected. This option does not apply to single line phones. By default, Auto Hold is enabled.

RESET REQUIRED!

Auto Page: Works in conjunction with AutoPage Time and AutoPage Zone. If AutoPage is enabled, the values set up in AutoPage Time and AutoPage Zone take effect. The default Auto Page Time is 10 seconds.

RESET REQUIRED!

Hands Free (only for DP200 display phones): When a user calls a DP200 display phone that has Hands Free enabled, the phone automatically connects to the call without ringing and is put on the speakerphone. A two-beep tone is heard before the call is put on the speakerphone to alert the station user. Hands Free only works for internal extension-to-extension calls. It does not work when transferring calls.

Exec Priv: Enables station (except Telecor Attendant stations) to monitor conversations on other extensions. This is a useful feature in evaluating call handling techniques and customer service. The extension monitoring the call can listen to both ends of the conversation, but cannot be heard. Both parties of a call being monitored hear a beep tone every 10-15 seconds to alert them that the call is being monitored. To use this feature a station user dials **# # 0** (or clicks the Monitor button in

Connect), dials the Monitor Password set in the System Configuration 2 and then dials the extension. *See System Configuration 2 on page 160 for complete setup of this feature.*

Can Be Monitored: Enables the extension to be monitored by extensions with the Exec Privilege feature enabled. Monitored parties hear a beep tone every 10-15 seconds while the call is being monitored. *See System Configuration 2 on page 160 for more information.*

Force Acct Codes: The Force Acct Codes feature requires a phone user to enter a code to place an outgoing non-emergency call. After a number is entered the user hears four beeps to prompt input of an account code. If a correct account code is *not* entered within 10 seconds, the call is terminated. If a correct account code is entered, the code is compared with the code in the existing list. If no match is found, the user hears four beeps again and can re-enter the account code. *See Forced Accounts on page 130 to set up codes.*

Disconnect Signal (only for standard phone and modem/fax): Any time an Extension port ends a call and the Disconnect Signal is enabled on that port, before returning to internal dial tone, the system sends "D" DTMF tone. The "D" tone is generally used when you have an Interactive Voice Response (IVR) or other third party equipment connected to an Extension port on the VS1 phone system. Third-party equipment needs to be configured to disconnect upon hearing a "D" DTMF tone. The Disconnect Signal setting works with the Gateway setting in a CO port. When Gateway is selected, and the system hears a "D" DTMF tone, it automatically disconnects the call.

Message Lamp (only for standard phone and modem/fax): If you are using a standard single line phone with an analog or 90V message waiting light, Message Waiting does not apply. The PEU now supplies 90V to the message waiting light for any port configured as a standard phone. This feature is for backward compatibility with older model phones.

Note	VS1 Software Versions 2.7.5 and later support analog message waiting lamps on ports configured as "Standard Phone" or "Modem/Fax." To support this feature, the system must have a Host Adapter Card Model 200 (Serial number begins with 318), with a PEU Model 200 or 205 connected to the Card. This feature is not supported if a PEU 250 is connected to the Host Adapter Card. If the PEUs are in a daisy-chain, the PEU must be the first in the chain for the message waiting lamp feature to work on that PEU. For more information, contact Telecor Technical Support.

Communication: Used for Attendant Stations only. Select the Communication port that Attendant connects to on the Telecor Voice Server (TVS). Enter 9600 for **Bits/Sec** and **COM 1** or **COM 2** depending on what serial port is being used.

PROGRAM PASSWORD

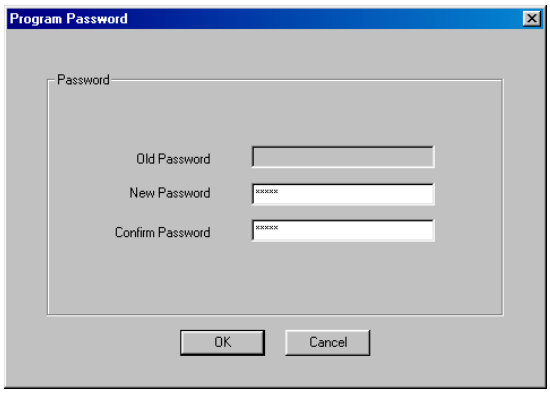
The Program Password enables you to specify a password that is used each time the VS1 Editor application is started. To create a Program Password, complete the following steps:

1. Select **Program Password** in the Tree Control Display.

   - The **Program Password** pane appears.

2. Click the **Add Password** button.

   - The **Program Password** dialog box appears.



Program Password dialog box

3. In the **New Password** text box type the Program Password. Program Passwords can be alphanumeric and have from 1-15 characters.

4. In the **Confirm Password** text box type the Program Password again.

5. Click **OK**.

Removing a Program Password

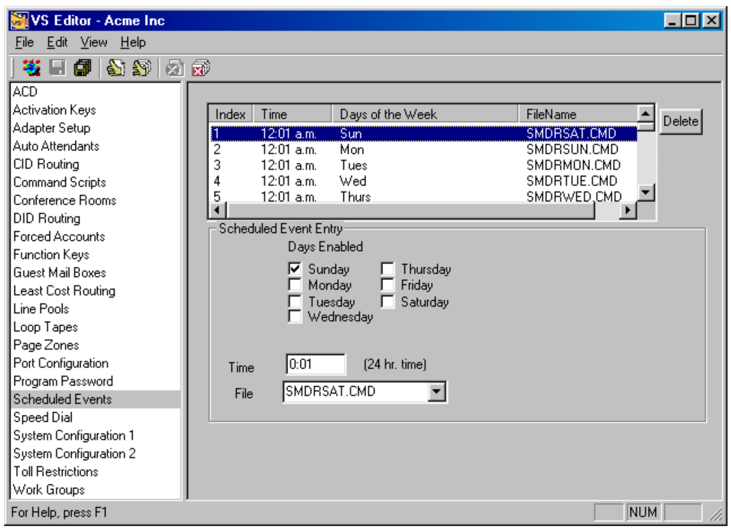
If you set up a Program Password and forget what it is, you must remove the password and create a new one. The Program Password can be removed either on-site or remotely. To remove the password remotely, you must connect with the Tel-Site application and through the Terminal window type `del cfg\cfg.ini` .

SCHEDULED EVENTS

A Scheduled Event runs a Command Script at a specific user-defined time during a day of the week. Up to 114 Scheduled Events can be created to run anytime during a weekly schedule (Sunday through Saturday). Scheduled Events run on a weekly repeating basis—not for specific days such as July 4, 2003. For example, every Tuesday at 3:00 p.m. you could run a Command Script that plays a message over the paging system.

The most common examples of Scheduled Event:

- Loading the Day or Night configurations.
- Playing prerecorded announcements over the paging system.
- Copying SMDR data to system files.
- Performing scheduled hard drive maintenance.



Scheduled Events pane

To create a Scheduled Event, complete the following steps:

1. Select **Scheduled Events** in the Tree Control Display.

   - The **Scheduled Event** pane appears. In the list box, there is a number of default Scheduled Events, including the Command Scripts that backup SMDR data to system files. Do not delete these Scheduled Events as they keep the original summary.dlm and acd.dlm output files (the files where SMDR data is collected) down in size.

2. In the list box select a Scheduled Event number line that is empty.

3. In the **Scheduled Events Entry** group box, enter the following information:

   - Check the days of the week that the Scheduled Event will run.
   - In the **Time** text box, type the time of day in 24-hour format that you want the Command Script to run.
   - In the **File** drop-down box, select the Command Script that will run.

4. Click the **Save** button in the toolbar.

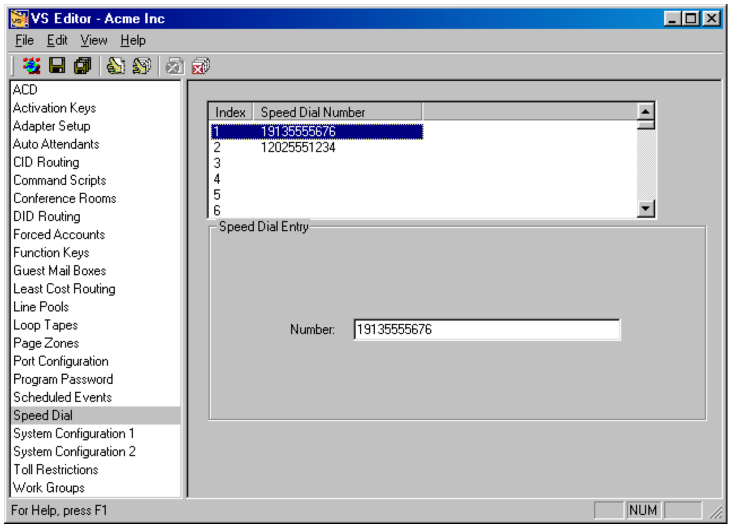
- The Scheduled Event is added to the list box above. The VS1 System puts Scheduled Events in order of the time of day they run. This helps you keep track of the events you have running during any period of time.

Note	Do not schedule multiple events to run at the same time during a day. Doing so might cause a Command Script to not run, and you cannot control which one will run first.

Note	Whenever the VS1 System is reset, the last Scheduled Event to run prior to the reset is run again. For example, if you schedule a system reset at 4:00 a.m., the system looks at the last Scheduled Event to run prior to 4:00 a.m. and then runs that event. For this reason, by default, the Night configuration is scheduled to run at 12:02 a.m. every day, one minute after the SMDR data is backed up. In case of a reset, this prevents the SMDR data from being backed up again and overwriting what data was collected for the day.

SYSTEM SPEED DIAL

The VS1 System provides 99 system-wide Speed Dial numbers that can be dialed from any VS1 System station option. Phone numbers are assigned to Speed Dial codes (01 through 99), which are entered by the station user to dial the phone number. After phone numbers are assigned to Speed Dial codes, a directory must be created and distributed to appropriate personnel at the installation site.



System Speed Dial pane

To create a System Speed Dial, complete the following steps:

1. Select **Speed Dial** in the Tree Control Display.

   - The **Speed Dial** pane appears.

2. Select a **Speed Dial** code number from the list box.

3. In the **Number** text box, enter the phone number for the Speed Dial code.

Note	You do not have to add a **9** to prefix your Speed Dial number. The speed dial numbers are always dialed using Dial Group 9.

4. Click the **Save** button in the toolbar.

   - The Speed Dial number appears in the list box above.

5. Repeat steps 2-4 for additional Speed Dial number or to edit an existing number.

Dialing System Speed Dials from a Station Option

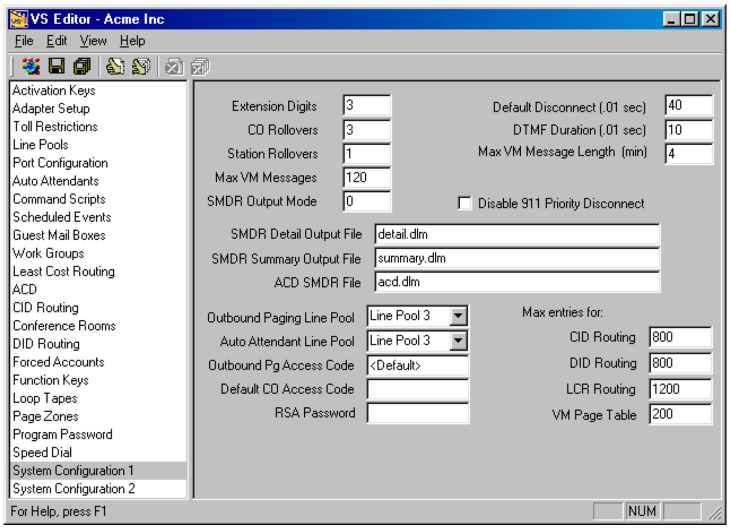
For VS1 Station Option users, dial ***4** and then the Speed Dial code number. DP200 Phone users have the alternate option of pressing the **System Speed Dial** Feature button instead of ***4**.

Note	Be certain to inform end-users that single number codes must be preceded by a 0 to provide a complete two-digit entry. For example, for the speed dial code of 08, the user would dial ***408.**

SYSTEM CONFIGURATION 1

RESET REQUIRED!

System Configuration 1 is a collection of system parameters. Generally the parameters are settings that have a system-wide effect, and are used for default values. The following is a brief explanation of the parameters and what should be entered in the text or drop-down boxes. Click the **Save** button in the toolbar when information has been changed or entered.



System Configuration 1 pane

Extension Digits: Determines the length of Extension numbers. The default is set to 3. Valid entries include 2, 3, or 4. Telecor strongly recommends the use of 3-digit Extension numbers. Extension Digits should only be configured during the initial system installation.

CO Rollovers: Determines the number of times a call to a CO port rolls over. The default is 3. This setting does not require adjustment.

Station Rollovers: Determines the number of times a call to a station port rolls over. The default is 1. This setting does not require adjustment.

Maximum VM Messages: This is system-wide setting for the number of Voice Mail messages each station's Voice Mail, or Guest Mailbox can hold. The default is 120. It can be reset to a minimum of 1 or to a maximum of 999. However, setting this to a large number can severely impact the port capacity of the system due to the memory allocation.

Default Disconnect (.01 sec): This is a system-wide setting for the CO disconnect signal. The default is 40 (400 milliseconds). Any drop in loop current on a CO line that is equal to or larger than this period of time results in a disconnect. Setting this value incorrectly can cause system errors. Do not change this value without first consulting VS1 Technical Support.

DTMF Duration (.01 sec): This is the system-wide setting for the length of time each DTMF tone is generated for dialing out on a CO port. The default is 10 (100 milliseconds). Consult VS1 Technical Support before changing this setting.

Max VM Message Length (min): This is the system-wide setting for the length of a Voice Mail message. The default is 4 minutes. The maximum length of time is 99 minutes.

SMDR Detail Output Mode: There are two valid options: **0** or **1**. The default is **0**, which sets the SMDR Detail Output Mode to unprotected. Setting the output mode to **1** means the file is protected.

Protected (1) must be selected when using third-party SMDR report software to prevent installation on other systems without license. This adds the last 5 digitis of the first HA serial number to the software protection field of the SMDR response.

Disable 911 Priority Disconnect: If a user dials 911 (or 9,911) and all CO ports are busy, the system will disconnect a non-emergency (911) call to expedite the connection. This is enabled by default.

SMDR Detail Output File: Type the file name for SMDR Detail output in this text box. (Telecor recommends **detail.dlm**). Any valid path and name can be entered. For example: **c:\ops\detail.dlm**. If no path is included the file is stored in the **c:\ops** directory. Detail files give one line of information per call action. SMDR Detail provides more detailed information for each call, breaking the call into segments and recording every change in action on a phone. It can be quite extensive, but is a good way to find out exactly what is happening with a phone.

SMDR Summary Output File: Type the file name for SMDR Summary output in this text box. The file is set to summary.dlm by default. Any valid path and name can be entered. For example: **c:\ops\summary.dlm**. If no path is included the file is stored in the **c:\ops** directory. SMDR Summary files give a summary of a call: where it rang in, date and time, length of call and so on.

ACD SMDR File: Type the file name for SMDR ACD output in this text box. The file is set to **acd.dlm** by default. Any valid path and name can be entered. For example: **c:\ops\acd.dlm**. If no path is included the file is stored in the **c:\ops** directory. ACD files give information about all ACD activities, including Queue Warning, Terminated Calls, Wrap-up and so on.

Outbound Paging Line Pool: A Line Pool must be specified here if Voice Mail Notification on a Digital Pager is to be used. The default is the **<Default>** Line Pool. Any valid Line Pool can be specified in this text box.

Auto Attendant Line Pool: A Line Pool must be specified here if the Transfer to Outside Line feature is to be used. The default is the **<Default>** Line Pool. The system searches through this Line Pool for a CO line to dial out on.

Outbound Paging Access Code: This feature is disabled on the VS1 System.

Default CO Access Code: This text box is used to specify an Access Code if the system is connected to Centrex lines that require a 9 or other DTMF tones before the phone number is dialed. This text box is blank by default. You need to add the word **<Default>** to the Extension Port Configuration under the **Access Code** text box for this feature to work properly.

RSA Password: This field determines the password for Remote System Access (RSA) which enables a modem to communicate with the system. The RSA Password is blank by default. Assign a password if a modem is connected to the Telecor Voice Server. The password can be alphanumeric. The RSA password entered in the **System Configuration 1** pane should match the **RSA password** text box in the **Connection** window of the site (accessed through Site Selection) when using the Tel-Site system management application. The **RSA Password** text box left blank means there is no password.

Max Entries For
The following text boxes are used to set the maximum entries for the respective items. These entries are used for better memory control usage by communicating to the system how much memory to allocate for each setting. It is recommended that these default settings not be changed.

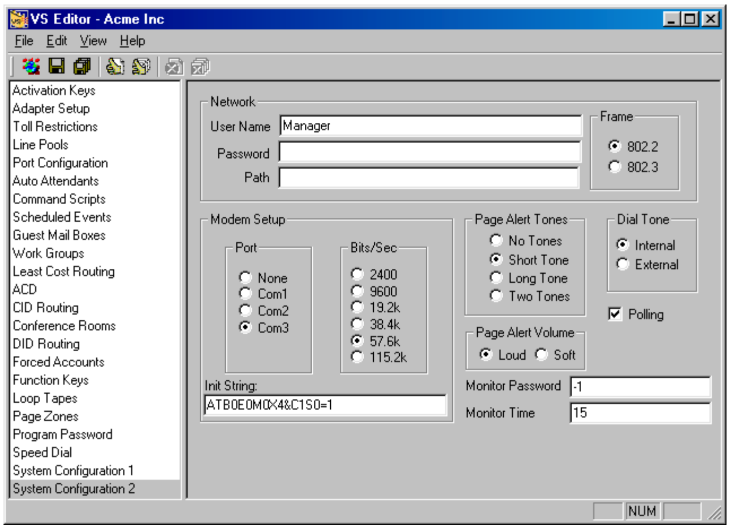
CID Routing: CID Routing is set to 800 by default.
DID Routing: DID Routing is set to 800 by default.
LCR: Least Cost Routing is set to 1,200 by default.

VM Page Table: Voice Mail Page Table is set to 200 by default. This is the number of page notifications saved by the system and the information from that action, such as the number of paging attempts, when to page again, and what number to dial.

SYSTEM CONFIGURATION 2

RESET REQUIRED!

System Config 2 is a collection of system parameters. The following is a brief explanation of the parameters and what should be chosen or entered in the accompanying text boxes. Click the **Save** button in the toolbar when information has been changed or entered.



System Configuration 2 pane

<u>**Network** Group Box</u>

Warning!	The information under **Network** is only valid for use if you are writing the SMDR information out through the network card. Otherwise, do not enter any information in these text boxes.

UserName: Type a valid username on the network. The UserName must have a password to work correctly. This is a case-sensitive text box.

Password: Type a valid password on the server for the user described in the **UserName** text box. This is a case-sensitive text box.

Path: Path is the server name, volume, and path that you want the SMDR information written to on your server. For example, **dsam/sys:support**

- This configures the network path to the support directory on the **sys** volume of the DSAM server.

Frame: Frame needs to be set to match the frame type of the network server that you are logging on to.

<u>**Modem Setup** Group Box</u>

Port: The internal modem Telecor puts in the TVS is set to COM 3 by default. The only time you would change this is if you remove the modem, or if you connect an external modem to a different COM port.

Baud Rate: The Baud Rate is set to 57.6 Kbps by default. The internal modem currently shipping in the TVS has a baud rate of 56 Kbps. The Baud Rate should be equal to or greater than the maximum speed of the modem in the TVS. This setting does not need adjustment.

Init String: The default modem initialization string in the **Init** text box is as follows: **ATB0EDMOX4&C1SO=1**. This setting requires adjustment if you replace the modem with one not supplied by Telecor. Compare the commands of the Telecor modem initialization string and the modem you want to use, and then adjust the **Init String** text box accordingly.

Dial Tone: The Dial Tone is set to Internal by default. The sound of the internal dial tone is heard when a phone user goes off-hook. If the Dial Tone is changed to External, it changes the sound of the dial tone – not the dialing process.

Polling: For Telecor Technical Support use only.

Page Alert Tone: Page Alert Tone determines which tone is played when a Paging Zone is activated. Paging Alert Tone is set to Short tone by default. Choose from None, Short tone, Long tone, or Two tones.

Page Alert Volume: Page Alert Volume is set to Loud by default. Choose either Loud or Soft Page Alert Volume.

Monitor Password: Select a five-digit password that extensions with Executive Privilege must enter to begin monitoring an extension.

Monitor Time: A beep tone is heard by all monitored parties every **{n}** seconds set in this text box. Ten seconds is the shortest interval between beep tones and 15 seconds is the longest interval permitted. Any value entered less than 10 results in a 10 second interval, while any value greater than 15 results in a 15 second interval between beep tones.

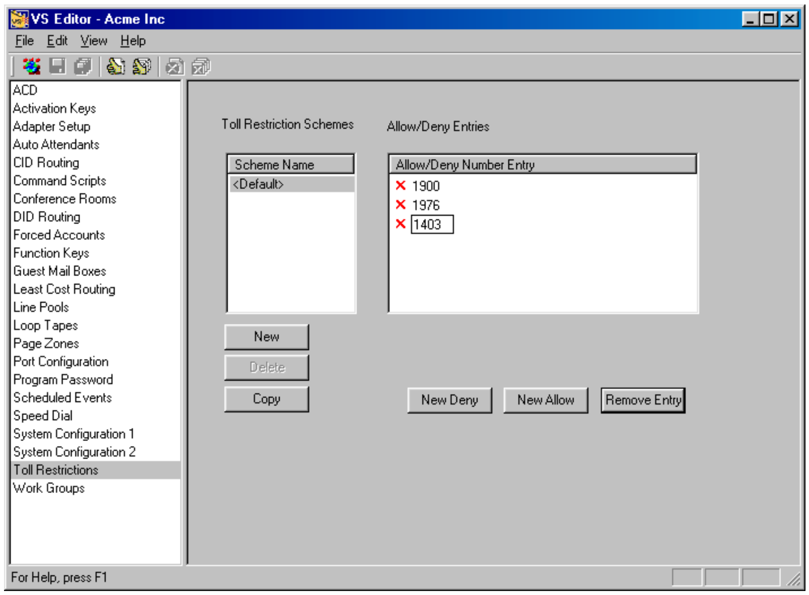
Note	By default, the beep tone cannot be disabled. However, under special circumstances the beep tone can be disabled with an activation key. Contact Telecor Technical Support for authorization requirements to obtain this activation key.

Note	To enable the Executive Privilege feature on the TVS you must edit the **`pbx`** line of the **`startpbx.bat`** file located in **`c:\ops`** to read **`pbx -e1`**. To disable the Executive Privilege feature after you have enabled it, edit the **`pbx`** line to read **`pbx -e0`**. The default setting for the Executive Privilege feature is off.

TOLL RESTRICTIONS

Toll Restrictions deny certain long distance calls being made. With the VS1 System, Toll Restriction Schemes are created which consist of a list of various numbers that cannot be made. The Scheme can also consist of specific dial strings that are allowed to go through that would otherwise be blocked. Once a Scheme is created it is then applied to an individual Extension in Port Configurations.

The VS1 Editor consists of a Default Scheme that denies 1900 and 1976 numbers. These calls are reverse charge calls and should not be removed.



Toll Restriction pane

Denying Additional Numbers in the Default Scheme

To deny additional numbers to the Toll Restriction Scheme, complete the following steps:

1. Select **Toll Restriction** in the Tree Control Display.
   - The **Toll Restriction** pane appears.
2. The **Scheme Name** list box consists of a **<Default>** Scheme. Click on this Scheme.
3. In the **Allow/Deny Number Entry** list box, the 1900 and 1976 numbers are listed with a red **X**, indicating that any dial string starting with these numbers is denied.
4. Click **New Deny**.
   - An empty dial string box marked with a red **X** appears in the **Allow/Deny Number Entry** list box.
5. Enter the dial string that will be denied. The VS1 System performs a "Best Match" type of comparison between the dial string and the contents of the defined Toll Restriction Scheme.

Therefore, if the dial string consists of 1403, all calls beginning with 1403 are denied. If the dial string consists of 14037, only numbers beginning with 14037 are denied and all other 1403 calls are allowed.

Note	The wildcard **x** can be used to represent a single digit, matching any number 0-9 in dial string. The **Toll Restriction Scheme** sorts dial strings in numerical order **0** through **9**, and then the wildcard **x**.

Note	If a number is dialed that does not match anything in the Toll Restriction Scheme, the call is allowed.

6. Click the **Save** button in the toolbar.

7. Repeat step 4-6 for other denied entries.

Creating a New Toll Restriction Scheme

To create a Toll Restriction Scheme, complete the following steps:

1. Select **Toll Restriction** in the Tree Control Display.

   - The **Toll Restriction** pane appears.

2. To add a new Scheme, click **New**.

   - The Scheme is added in the **Scheme Name** list box above and is numbered accordingly.

3. Select the new Scheme and type a name for it.

4. Click **New Deny**.

   - An empty dial string box marked with a red **X** appears in the **Allow/Deny Number Entry** list box.

5. Enter the dial string that will be denied. The VS1 System performs a "Best Match" type of comparison between the dial string and the contents of the defined Toll Restriction Scheme. Therefore, if the dial string consists of 1403, all calls beginning with 1403 are denied. If the dial string consists of 14037, only numbers beginning with 14037 are denied and all other 1403 calls are allowed.

Note	The wildcard **x** can be used to represent a single digit, matching any number 0-9 in dial string. The **Toll Restriction Scheme** sorts dial strings in numerical order **0** through **9**, and then the wildcard **x**.

Note	If a number is dialed that does not match anything in the Toll Restriction Scheme, the call is allowed.

6. Click the **Save** button in the toolbar.

7. Repeat step 4-6 for other denied entries.

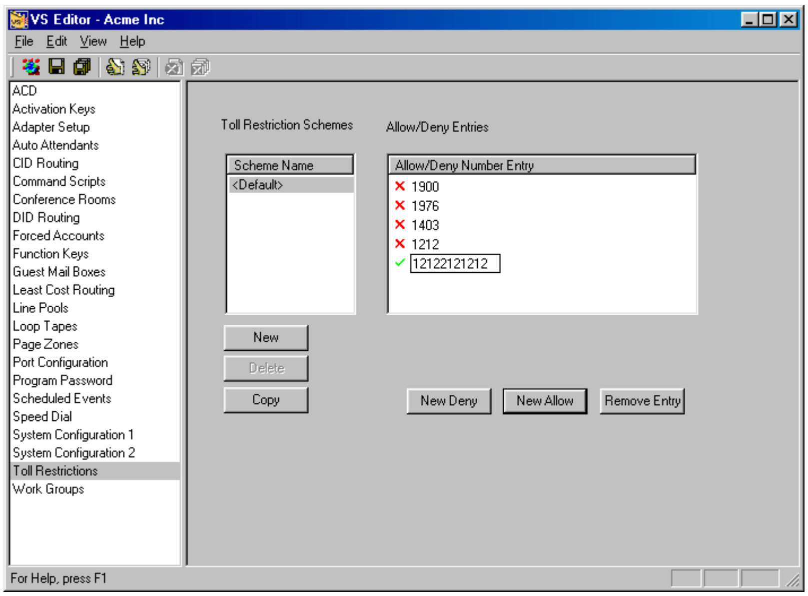
Allowing Entries in a Toll Restriction Scheme

To allow certain dial strings to go through that would otherwise be blocked in the Toll Restriction Scheme, complete the following steps:

1. With the Toll Restriction Scheme selected, click **New Allow**.

2. A dial string box marked with a green check mark appears in the **Allow/Deny Number Entry** list box.

3. Enter the dial string that will be allowed. For example, if the number 1-212-212-1212 needs to called in a Scheme that denies all dial strings beginning with 1212, enter 1-212-212-1212 in the dial string box.



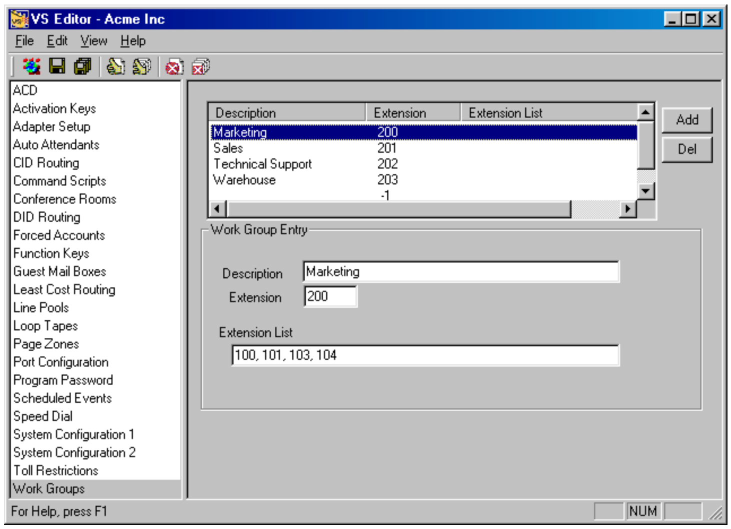
Toll Restriction pane

Toll Restriction Schemes are activated in each individual Extension port type configuration. *See "Setting up an Extension Port Type" on page 149 for more information.*

WORK GROUPS

RESET REQUIRED!

Work Groups on the VS1 Telephone System enable you to assign a virtual extension to a group of stations. When the Work Group extension is called, all extensions in the Work Group ring simultaneously. There is a maximum of 20 Work Groups.



Work Group pane

To create a Work Group, complete the following steps:

1. Select **Work Groups** in the Tree Control Display.

   - The **Work Group** pane appears.

2. Click **ADD**.

3. In the **Work Group Routing Entry** group box, enter the following in the corresponding text boxes:

   Description: Enter a name or description for the Work Group that will appear on DP200 display phones and CTI station options.

   Extension: Enter an extension for the Work Group.

   Extension List: Type the extensions you want included in the Work Group. This can also includes Guest Mail Boxes. Up to 20 extensions can be listed in this text box. Use commas to separate the extensions. Do not use spaces between the extensions. For example, **101,102,103,104,105**.

Note	The first valid station listed in the **Extension List** text box in the **Work Groups** window controls the Rollover Port and the Rollover Time for that Work Group. If the first valid station has a Rollover Port configured to Voice Mail, then unanswered calls to the Work Group rollover to that extension's Voice Mail.

4. Click the **Save** button in the toolbar.

   - The Work Group entry is added to the list box above.

5. Repeat steps 2-4 for other Work Group entries.

Note	Messages can be forwarded to the Work Group extension and each physical extension and Guest Mailbox in the Work Group will receive a copy of the message.

VALIDATING THE CONFIGURATION

The VS1 Editor is capable of informing you of potential errors while configuring a VS1 system. Errors are listed for a specific pane. In order for potential errors to be documented the pane must first be validated. Validation can be conducted on a single pane or all panes.

Validating a single pane can be accomplished by:

- Switching between one pane to another in the Tree Control.

- Pressing the **Save this Pane** button in the toolbar.

- Pressing the **Validate this Pane** button in the toolbar.
- Selecting **Edit > Validate > This Page** in the menu bar.

Validating all panes can be accomplished by:

- Pressing the **Save All Panes** button in the toolbar.

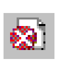
- Pressing the **Validate all Panes** button in the toolbar.
- Selecting **Edit > Validate > All Pages** in the menu bar.

Displaying Errors

Once a pane has been validated, any potential error messages for it can then be displayed. There are a number of ways that error messages can be displayed.

- Pressing the **Validate this Pane** button validates the pane and then displays any error messages for the pane.

- Pressing the **Validate all Panes** button validates all panes and then displays the **Validation Results** window, where any panes with errors are listed. Select a pane and click **Select** to display its error messages.



- Pressing the **Errors on this Pane** button displays any errors for the currently selected pane.

- Pressing the **Errors on all Panes** button displays the Validation Results window, where any panes with errors are listed. Select a pane and click **Select** to display its error messages.
- Selecting **View > Display Errors > This Pane** from the menu bar displays errors on the currently selected pane.
- Selecting **View > Display Errors > All Panes** from the menu bar displays the **Validation Results** window, where any panes with errors are listed. Select a pane and click **Select** to display its error messages.

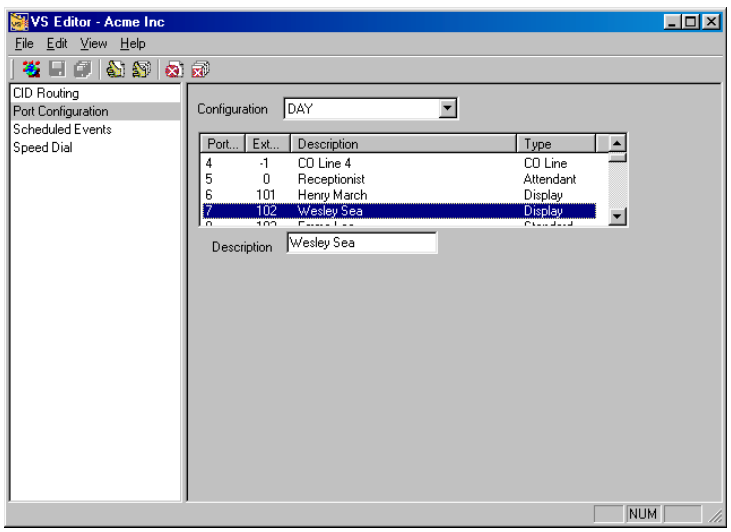
VS1 EDITOR – LIMITED FEATURE VERSION

The VS1 Editor can be installed with limited features to allow end-users the ability to make common changes to the VS1 system without risk of modifying or damaging important configuration data. Installing the VS1 Editor with limited features gives end-users the ability to:

- Set up Caller ID Routing in the CID Routing pane.
- Change descriptions of ports in the Port Configuration pane.
- Change time and day of weeks for command scripts in the Scheduled Events pane.
- Set up system speed dials in the Speed Dial pane.

To provide a customer site with a limited set of VS1 Editor features, install the application onto the customer computer as you would normally, but ***do not enter the licensing information when prompted, which includes name, company and registration number!***

In addition, ensure that Tel-Site is also installed, and that an on-site connection is set up for the customer to implement any changes (*see "Connection Methods" in Tel-Site section*).



Port Configuration pane of VS1 Editor installed with limited features.

Station Options

STATION OPTIONS OVERVIEW

This section describes the station options available for use with the VS telephone system. Station options are classified into two groups: Computer Telephony Integration (CTI) applications, and Phones. The features of each option are briefly described, along with system requirements and a quick guide to basic steps.

For detailed instructions on the use and installation of each station option, please refer to its separate User's Guide.

The CTI applications described in this section include:

- Telecor Attendant CTI client application
- Telecor Connect CTI client application

The Phone options described in this section include:

- Display Phone Model 200 (DP200)
- HS-1301-LN Telephone (HS-1301-LN)

Also described in this section are Feature Keystrokes used with the Telecor VS1 phone system, an overview of the Voice Mail features built into the system, and end-user instructions for gaining access to and customizing Voice Mail features for station options.

Overview of Attendant

The Attendant CTI client application for the Windows® operating system is a powerful primary answering position for a business with high call activity. Attendant improves the ability to handle calls and work in other Windows applications between calls. Three call-processing windows—**Calls**, **Extensions**, and **Transfer**—provide simple and efficient call handling with a minimal amount of training or experience.

The **Calls** window indicates incoming calls. If Caller ID is installed on the VS1 telephone system, a **Call** status panel provides caller information. Call handling buttons simplify and speed operations.

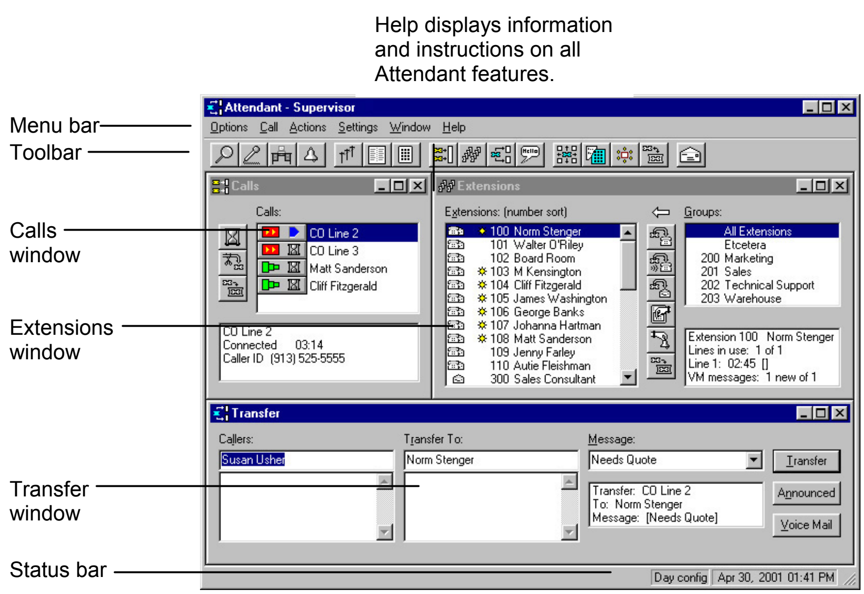
When calls arrive in the **Calls** window, they can be transferred to the **Extensions** window using the drag-and-drop feature. The **Extensions** window provides a directory of extensions and groups, and displays the current status of each.

The **Transfer** window interfaces with a Caller Database to assist in transferring calls. The Message Feature enables the user to screen calls and transfer a text message to station displays with each call.

In addition to basic call processing functions, Attendant provides powerful “blind operations,” enabling calls to be selected in the **Calls** window and processed without actually connecting to the call, increasing call processing speed. Attendant also enables the user to monitor and transfer calls to Auto Attendants, ACDs, Conference Rooms, and Park Zones, merge calls, and make paging announcements.

Attendant integrates seamlessly into a user’s work habits, providing a variety of ways to process calls. Users can use the mouse, or keyboard and keyboard shortcuts, or a combination of both to handle calls. Attendant works the way the user does.

The Attendant application is a Windows®-based program that operates on Microsoft® Windows® 98, Windows® 2000, Windows® Me, and Windows® XP operating systems. Installing a Computer Telephony Interface Module (CTIM) integrates the telephone into the computer, and allows Attendant to be used with a headset.



Attendant Main window

Product Features

- Automatic Call Distribution (ACD) Group information
- Away From Desk option
- Caller Database
- Caller ID
- Call Merging
- Conference Room information
- Display Messaging
- Drag-and-Drop features
- Extension Status Review
- Greeting window
- Parking Zones window
- Paging Zones window
- Phonebook
- Resizable and configurable layout
- Voice Mail

Requirements

- Pentium 120 PC or faster, 32 megabyte (MB) of RAM, 1 gigabyte (GB) hard drive
- Microsoft® Windows® 98, Windows® 2000, Windows® Me, and Windows® XP operating systems.
- Open COM port with dedicated IRQ
- 15-inch color monitor recommended; 640x480 mode
- 1312 kilobytes (K) of available disk space
- CTIM and Headset

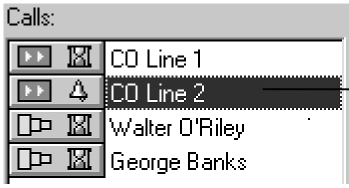
Includes (Part Number: PVS-SCT-A01)

- Telecor Attendant CD
- Attendant Activation Key Request Form
- Two RJ11-DB9F connectors

Using Attendant—Basic Operation

Selecting a Call

1. Click the text portion of the call description in the **Calls** list of the **Calls** window.



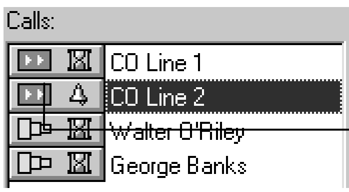
Click call description to select a call

Selecting a call

Connecting to a Call

1. Click the **Call** button next to the call description in the Calls list of the Calls window.

   - Connecting to a second call automatically places a currently connected call on hold.



Click the Call button to connect a call

Connecting to a call

Placing a Call On Hold

1. In the **Calls** window, click the call description to select a call, or click the Call button to answer a call.

2. Click the **Hold** button in the Calls window.

   - The call status symbol changes to the hourglass.

Taking a Call Off Hold

1. Click the **Hold** status button next to the call description in the **Calls** list of the **Calls** window.

Disconnecting a Call

1. In the **Calls** window, click the call description to select a call, or click the **Call** button to answer a call.

2. Click the **Disconnect** button in the **Calls** window.

Transferring a Call to an Extension Unannounced

1. In the **Calls** window, click the call description to select a call, or click the **Call** button to answer a call.

2. Click the extension in the **Extensions** list of the **Extensions** window to which you want to transfer the call.

3. Click the **Transfer** button in the **Extensions** window.

Transferring a Call to an Extension Announced

1. In the **Calls** window, click the call description to select a call, or click the **Call** button to answer a call.

2. Click the extension on the **Extensions** list in the **Extensions** window to which you want to transfer the all.

3. Click the **Announced Transfer** button in the **Extensions** window.

4. Announce the caller to the extension, then click the **Disconnect** button.

Transferring a Call to an Extension's Voice Mail

1. In **Calls** window, click the call description to select a call, or click the **Call** button to answer a call.

2. Click the extension on the **Extensions** list in the **Extensions** window for the Voice Mail to which you want to transfer the call.

3. Click the **Transfer to VM** button in the **Extensions** window.

Making an Outside Call

1. Click the **Outside Call** button on the toolbar.

2. Type the number you want to call in the **Number** text box of the **Outside Calls** dialog box. You do not need to type a **9** prior to the number.

3. Click **Dial**.

- Click the **Keypad Dialer** button on the toolbar if you need to dial additional numbers after you are connected.

Making an Inside Call

1. Click the **Inside Call** button on the toolbar.

2. Select the extension from the **Extensions** list, or type in the extension number in the **Number** text box of the **Inside Calls** dialog box.

3. Click **Dial**.

For more detailed instructions on the use and installation of the Attendant CTI application, please refer to the *Telecor Attendant CTI Application User's Guide*.

Overview of Connect

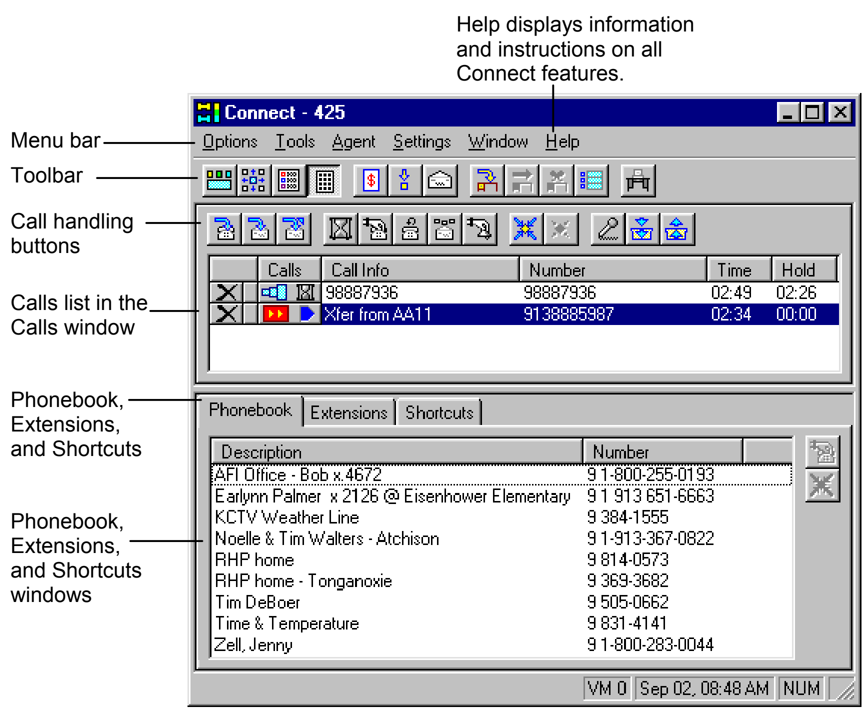
The Connect CTI client application for the Windows® operating system removes the phone from the desk and puts it inside the computer. Installing a Computer Telephony Interface Module (CTIM) integrates the telephone into the computer. Installing a PC Option Module (PCOM) enables a user to use the computer screen for call processing and keep a phone on the desk. Keeping a phone on the desk also allows calls to be received when the computer is turned off. Connect integrates seamlessly into a user's work habits, providing a variety of ways to use the computer to process calls. A user can use the mouse, keyboard and keyboard shortcuts, or a combination of both to handle calls. Connect works the way the user does.

The **Calls** window indicates incoming calls. If Caller ID is installed on the phone system, that information is displayed in the **Calls** window. Call handling buttons simplify and speed operations. In addition to the **Calls** window, Connect includes **Phonebook**, **Extensions** and **Shortcuts** tabs that a user can customize to meet his or her calling needs. Features such as Blind Transfer, Blind Hold, Account Codes, Station Status, and many others are available at the click of a mouse button.

The **Call Bar** window enables calls to be processed with a scaled-down version of the Connect main screen. The Connect **Call Bar** window can be customized to pop-to-the-top when a call arrives while working in another application.

A user can also monitor Automatic Call Distribution (ACD) with Connect, and track individual ACDs, the number of agents logged on, and the number of calls in the queue.

The Connect application is a Windows®-based program that operates on Microsoft® Windows® 98, Windows® 2000, Windows® Me, and Windows® XP operating systems, and supports Dynamic Data Exchange (DDE) clients and servers. Telecor Connect can be used with either a headset or a handset.



Connect Main window

Product Features

- Automatic Call Distribution (ACD) Monitor window
- Account Code Tool
- Blind Hold with Message
- Blind Transfer
- Bring to Top on Ring
- Call Bar
- Call Processing Buttons
- Caller ID
- Dynamic Data Exchange (DDE) support
- Extensions List
- Online Help
- On-screen Keypad Dialer
- Personal Phonebook
- Shortcuts Buttons
- Simultaneous Call Handling (10 Calls)
- Station Status window
- TAPI support
- Voice Mail Message Indicator

Requirements

- Pentium 120 PC or faster, 32 megabyte (MB) of RAM, 1 gigabyte (GB) hard drive
- Microsoft® Windows® 98, Windows® 2000, Windows® Me, Windows® XP operating systems
- Open COM port with dedicated IRQ
- CTIM and Headset or PCOM and Single Line Phone

Includes (Part Number: PVS-SCT-T01)

- Telecor Connect CD
- CTI Activation Key Request Form

Using Connect—Basic Operation

Making a Call

1. In the **Calls** window, click the **Dial** button.

2. In the **Dial** dialog box, type the extension or phone number. If you are calling an outside phone number you must type a **9** at the beginning of the number.

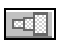
   - If you are dialing an internal number you can also enter an alphanumeric message in the **Dial** dialog box **Message** field. This message appears only on stations with a display, such as the Display Phone Model 200 (DP200) or an Attendant or Connect CTI application. Messages do not appear on external calls.

3. Click **Dial**.

Disconnecting a Call

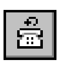
1. In the **Calls** window, click the **Disconnect** button in the **Calls** list.

Answering a Call

1. In the **Calls** window, click the **Internal Call** or **External Call** button in the **Calls** list.

Redialing a Number

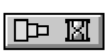


1. In **Calls** window, click the **Redial** button.

2. In the **Redial** dialog box, click to select a number from a list of earlier calls.

3. Click **Redial**.

Placing a Call On Hold



1. In the **Calls** window, click the **Call** button in the **Calls** list. The connected call graphic changes to an hourglass.

   Or

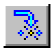
2. Click the **Hold** button.

Transferring a Call - Announced

1. In the **Calls** window, click the **Transfer** button.

2. In the **Transfer** dialog box, type the extension.

3. Click the **Transfer** button in the dialog box.



4. The extension to which you transfer the call rings (or connected immediately if Handsfree is enabled). Wait for the person to answer and announce the transfer. Then click the **Complete Transfer** button (see graphic to left). This is in the same location as the **Transfer** button. It changes appearance for an announced transfer.

Transferring a Call - Unannounced

1. In the **Calls** window, click the **Transfer** button.

2. In the **Transfer** dialog box, type the extension.

3. Click the **Transfer** button in the dialog box.

4. The extension to which you transfer the call rings. At this point, click the **Complete Transfer** button (this is the same button as the **Transfer** button).

Transferring a Call to Voice Mail

1. In the **Calls** window, click the **Transfer to VM** button.

2. In the **Transfer to Voice Mail** dialog box, type the extension of the Voice Mail to which you want to transfer the call.

3. Click **Transfer**.

Transferring a Call to Your Voice Mail

1. In the **Calls** window, click the **Transfer to my VM** button.

For more detailed instructions on the use and installation of the Connect CTI application, refer to the *Telecor Connect CTI Application User's Guide*.

Overview of Display Phone Model 200 (DP200)

The Display Phone Model 200 (DP200) is designed for use with the VS1 telephone system. The DP200 Display Phone has the capability to handle five calls. Three of the five **Line Appearance** buttons can be customized as **Extension View** buttons known as Direct Station Select/BusyLamp Field (DSS/BLF). Each of the five buttons provides a visual display of a call's status.

The DP200 Display Phone is equipped with a data port that enables users to plug in any 900 megahertz cordless phone or a modem. When a station receives a call, both the cordless phone and the DP200 ring; or plug a modem into this port and a computer can process data.

Using the program switch, the DP200 Display Phone can be customized for office settings or warehouse settings. In Office mode, settings for Page Volume, Ring Select and Ringer Volume are lower in volume than they are in Warehouse mode.

Five standard Feature buttons—Mute, Flash, Voice Mail, Redial, and Hold—and 12 customizable Feature buttons are included on the DP200 Display Phone. Custom Feature buttons enable users to program the features used the most by dialing a short code and following the voice prompts. During phone operation, the button lights up or flashes to inform the user that the programmed feature can be utilized or accessed.

The DP200 Display Phone is equipped with a 2 line by 16 character display, which provides useful information about the calls being handled. When a call is received the display changes to show information about the caller. The display shows what line the call is ringing on and whether it is an external or internal call. The A Feature button enables users to switch between name and extension. If the receptionist uses the Telecor Attendant CTI client application, pressing the A Feature button switches between the Attendant Message and the Caller ID number and name displays.

Pressing the B Feature button once shows the current status of some of the features such as Do Not Disturb, Call Forwarding, Handsfree and Call Waiting. Pressing the B Feature button twice enables users to view the status of their Voice Mail. Pressing the B Feature button three times shows the Automatic Call Distribution (ACD) display.

Telecor Display Phone Model 200 (DP200)

Product Features

- Auto Hold
- Auto Line Select
- Blind Hold with Message
- Blind Transfer
- Caller ID
- Conferencing
- Cordless Phone Support
- Discreet Call Screening
- Distinctive Ring
- Do Not Disturb
- DSS/BLF Capability
- Flash
- Handsfree Operation
- Headset Mode
- Hearing Aid Compatible
- Hold
- Large LCD Display
- Message Waiting Light
- Mute
- Office/Warehouse Mode
- Personal Speed Dial
- Redial
- Release
- Ringer Volume
- Simultaneous Call Handling (5 calls)
- Speakerphone
- System Speed Dial
- Transfer
- Twelve Customizable Feature Buttons
- Visual Park
- Volume Control
- Wall Mount

Using the DP200—Basic Operation

Making an Internal Call

1. Lift the handset.

2. Dial the extension you want to call.

Making an External Call

1. Lift the handset.

2. Press **9** for an external line.

3. Dial the phone number.

Answering a Call

1. Lift the handset.

   - Press the A Feature button to display Caller ID information.

Answering Multiple Calls

If you are using LINE 1 when another call comes in, a two-beep call waiting tone sounds (presuming Call Waiting is enabled).

1. Press the LINE 2 button one time to preview the call. Press the LINE 2 button a second time to answer the call and place the first call on hold.

Transferring a Call

After answering the call:

1. Press the TRANSFER Feature button and listen for the tone.

2. Dial the extension to which you want to transfer the call. Hang up to transfer.

Placing a Call on Hold

1. Press the HOLD Feature button once to place a call on hold.

Taking a Call off Hold

1. Press the flashing line button of the call on hold.

Redialing the Last Number Called

1. Press the REDIAL Feature button.

Changing the Number of Lines Available

1. Lift the handset and press **7801**.

2. During the Feature Programmer announcement, press **677**.

   - Do not wait for the announcement to end before pressing **677**.

3. Press **2** for a two-line phone; **3** for a three-line phone; **4** for a four-line phone; **5** for a five-line phone. Hang up.

To use the additional lines, press the appropriate Line Appearance button prior to dialing.

For more detailed information on the use and installation of the DP200 Display Phone, please refer to the *Telecor Display Phone Model 200 (DP200) User's Guide*.

DP200 Programmable Function Codes

The following list contains all user-programmable function codes for customizable feature buttons on a DP200 Display Phone.

Note You must dial a zero if the second digit in the code is preceded by zero. For example, to customize a Feature button for **Pickup Group (03)**, press **0** and then **3**.

Feature	Code	Description
ACD Log Off	**17**	Log off all ACD groups into which you are logged.
ACD Log On with Specific Code	**19** + **ACD code**	Log on to an ACD using a pre-programmed account code.
ACD Wrap-Up Cancel	**42**	Cancel the automatic wrap-up mode so that you can receive more calls from the ACD queue.
Blind Hold with Message	**26**	Place an incoming call on hold without first answering the call. A message plays stating, "You have been placed on hold by the person you are trying to reach. They have been notified of your call, and will be right with you."
Blind Transfer	**05** + **Extension**	Transfer a call ringing at your extension to another extension without connecting to the call.
Blind Transfer to Voice Mail	**40**	Transfer a call ringing at your extension to your Voice Mail without connecting to the call.
Call Waiting ON/OFF Toggle	**06**	Turn call waiting tones ON or OFF.
Conference	**44**	Conference with up to two other callers (internal or external) without using a Conference Room.
Dial	**07 + Phone/Extension number**	Dial a specific number. If programming an outside number, include a 9 and appropriate long distance digits prior to the number.
Do Not Disturb ON/OFF Toggle	**08**	Turn ON or OFF the Do Not Disturb setting.
Flash External	**75**	Used for Centrex transfers
Forwarding Cancel	**09**	Cancel any call forwarding that you previously programmed at your extension.
Forward From	**10**	Forward incoming calls received at another extension to your extension.
Forward From Specific Extension	**11** + **Extension**	Forward calls at a specific station to your extension.
Forward To	**12**	Forward your calls to another extension.
Forward To Specific Extension	**13** + **Extension**	Forward your calls to a specific extension.
Handsfree	**14**	Turn ON or OFF the Handsfree mode.
Message Light Off	**22**	Turn the Message Waiting Light OFF at an extension.

Feature	Code	Description
Message Light Off Specific Extension	**23** + **Extension**	Turn the Message Waiting Light OFF at a specific station.
Message Light On	**20**	Turn the Message Waiting Light ON at an extension.
Message Light On Specific Extension	**21** + **Extension**	Turn the Message Waiting Light ON at a specific extension.
Music Over Speaker	**39**	Hear music over the phone speaker, if your system is connected to Music Source 1.
Page Over Specific Zone	**25** + **Zone Number**	Access a specific Paging Zone to make an announcement.
Page Over Zone	**24**	Gain access to the Paging feature.
Park	**27**	Park the current call.
Park Retrieve	**28**	Retrieve a parked call.
Personal Speed Dial	**38**	Gain access to the Personal Speed Dial feature.
Pickup Extension	**01**	Answer a call ringing at another extension.
Pickup Group	**03**	Answer a call ringing at any phone in your group.
Pickup Specific Extension	**02** + **Extension**	Answer a call ringing at a specific extension.
Relay 1	**29**	Activate the dry contact for Relay #1 on the Port Expansion Unit. For example, the contact could be connected to a security door lock.
Run Specific Command File	**32** + **Command File Number (01 to 99)**	Run a system command file.
System Speed Dial	**33**	Gain access to System Speed Dial numbers.
System Speed Dial Specific Number	**34** + **Code**	Access a specific System Speed Dial number.
Station Status	**16**	Hear the system port number and extension number of the station.
Transfer	**35**	Transfer the call to another extension.
Transfer to Specific Extension	**36** + **Extension**	Transfer a call to a specific extension.
Voice Mail	**37**	Gain access to your Voice Mail.

Overview of the HS-1301-LN Telephone

The HS-1301-LN Telephone is designed for use with the VS1 telephone system. The HS-1301-LN Phone incorporates a standard 12-button touchpad dial, along with three special function buttons (store, recall, and save), a flash button, redial button, and three one-touch priority dial buttons. Calls from the HS-1301-LN are initiated by standard dialing procedures or by using the 10 number programmable auto-dialing features, or the three priority dial buttons.

Product Features

- Message Waiting Light
- 3 Special Function Buttons
- Flash Button
- Redial Button
- 3 Priority Dial Buttons
- 10 Number Speed Dialing
- 13 Memory Locations
- Ringer Volume Control
- Last Number Redial
- Desk or Wall Mount
- Ringer Volume
- System Speed Dial

Telecor HS-1301-LN Telephones

Using the HS-1301-LN—Basic Operation

Making an Internal Call

1. Lift the handset.
2. Dial the extension you want to call.

Making an External Call

1. Lift the handset.
2. Press **9** for an external line.
3. Dial the phone number.

Answering a Call

1. Lift the handset.

Transferring a Call

1. Press the FLASH button.
2. Press ***** and **7**.
3. Dial the extension to which you want to transfer the call.
4. Hang up.

Placing a Call On Hold

1. Press the FLASH button for System Hold.

   Or

2. Press the HOLD button.

Taking a Call Off Hold

1. Press the FLASH button to take a call off System Hold.

   Or

2. Press the HOLD button again to take a call off hold.

Redialing the Last Number Called

1. Press REDIAL button.

Quick Guide to VS1 Phone System Feature Keystrokes

You can use any of the following features on VS1 phones by using the appropriate keystrokes on the phone keypad.

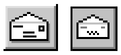
Feature	Enter the Keystroke Sequence
ACD Log Off	**7001**
ACD Log On	**7000 + ACD I.D.**
ACD Wrap-Up Cancel	**7002**
Call Forwarding to an Extension	*** 3 + Extension**
Call Forwarding–Cancel	*** * 3**
Call Forwarding to your Voice Mail	**# # 3**
Call Waiting Tone - Disable.	***5**
Call Waiting Tone - Enable	****5**
Direct External Dial Tone - Disable	**# # 4** + hang up
Direct External Dial Tone - Enable	**# # 4** + hang up
Do Not Disturb - Start	**# 5**
Do Not Disturb - Cancel	**# # 5**
Extension Status	**# # 7**
External Flash	**FLASH** + *** 2**
Message Light On	*** 6**
Message Light Off	**# 6**
Message Light On - Extension	*** * 6 + Extension**
Message Light Off - Extension	**# # 6 + Extension**
Page - All	*** * 9**
Page - Zone 1	*** 9**
Page- Zone 2	**# 9**
Park - Place	**FLASH** + **#** + **0-9**
Park - Retrieve	*** 2** + **0-9**
Pickup a Ringing Extension	**# 8 + Extension**
Pickup a Ringing Extension in Group	*** 8**
Pickup Line {*n*} from Cordless Phone[2]	**FLASH** + **# #** [**n**]
Redial	*** 0**
Reset Extension	*** #** + **8 6**
Speed Dial - Personal	**# 4** + **0-9**
Speed Dial - System	*** 4** + **01-99**
Transfer–Announced	**FLASH** + *** 7** + **Extension**
Transfer–Unannounced	**FLASH** + *** 7** + **Extension + hang up**
Voice Mail–leave message	**6** + **Extension**
Voice Mail–check	**6** + **your Extension**

[2] Feature only works with Display Phone.

VOICE MAIL ON THE VS1 PHONE SYSTEM

The VS1 telephone system provides you with powerful Voice Mail features. To use the Voice Mail features, use the **Keypad** on the phones, or the **Keypad** dialog box on the toolbar of the CTI applications.

Gaining Access to Your Voice Mail with Telecor Attendant or Connect



1. Click the **Voice Mail** or **My voice mail** button on the toolbar.

   - If prompted, click your passcode on the Keypad dialer.

Gaining Access to Your Voice Mail with the DP200

1. Lift the handset and press the VOICE MAIL button.

   - If prompted, dial your passcode.

Gaining Access to Your Voice Mail with the HS-1301-LN

1. Lift the handset and press **6** + **your Extension** number.

   - If prompted, dial your passcode.

The Main Voice Mail Prompts

After accessing Voice Mail, an announcement tells you how many new and old Voice Mail messages you have. The voice prompts lead you through the main Voice Mail features.

Note The steps you are asked to perform to use the Voice Mail features are documented with the mouse interface on the Keypad dialog box of the CTI applications. The voice prompts ask you to *press* a number; this document asks you to *click* a number on the Keypad dialer. If you are using a phone, press the appropriate key on the Keypad.

- Click **1** to listen to your new messages. After listening to new messages, you can listen to old or saved messages. *See "Listening to a Voice Mail Message," page 189.*

- Click **2** to gain access to the **Setup Options** menu. *See "Setting Up Voice Mail Features," page 190*.

- Click **7** to send a message or forward a Voice Mail message to one or more extensions. See *"Sending a Message to an Individual or Group," page 191* or *"Forwarding a Voice Mail Message to an Individual or a Group," page 192*.

- Click ***** to exit your Voice Mail.

Using the Keypad Dialer for Voice Mail Features

Use the Keypad to handle your Voice Mail calls the way you want. Figure 10 illustrates how to use the Keypad.

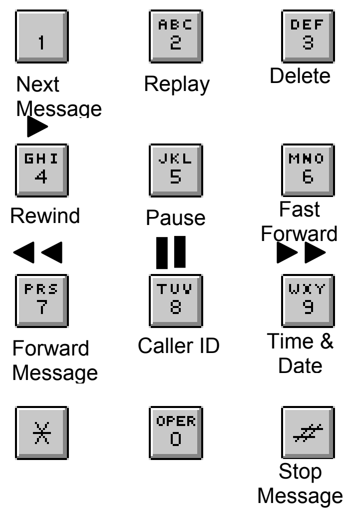


Figure 10
Keypad Voice Mail Functions

Listening to a Voice Mail Message

The Keypad can act similarly to the controls of a tape recorder while listening to your Voice Mail.

1. Gain access to your Voice Mail.
   - If prompted, click your passcode.
2. Click **1** to listen to a message.
   - The voice prompts offer nine options to choose from when the message is done playing.
   - Click **1** to play the next message.
   - Click **2** to replay the message.
   - Click **3** to delete a message. The message is erased from the phone system after a set number of hours, determined by the system installer.

Note If you don't delete your Voice Mail messages they are automatically saved. Check with your system installer to determine how long messages remain saved.

   - Click **4** to rewind the message for 5 seconds.
   - Click **5** to pause during the message (the message resumes in 20 seconds).
   - Click **6** to fast forward the message 5 seconds.
   - Click **7** to forward the message to another Voice Mail extension (you can record comments about the message you forward). For more information, *"Forwarding a Voice Mail Message to an Individual or a Group," page 192*.
   - Click **8** to hear Caller ID.

Note	Caller ID numbers are played for external calls only if that option is set up on the system.

3. Click * to exit your Voice Mail.

Setting Up Voice Mail Features

The following is a list of the **Voice Mail Setup Option**s menu.

1. Gain access to your Voice Mail.
   - If prompted, click your passcode.
2. Click **2** to gain access to the **Setup Options** menu. Voice prompts are given for each option.
   - Click **1** to record a new greeting. When you have finished recording your greeting, click **#** to stop. Click **2** to play back your recording.
   - Click **3** to create or change your passcode. Click the four numbers of your new passcode.
   - Click **4** to turn on or off the time and date notation with each message you retrieve.
   - Click **5** to turn on or off the Caller ID announced after each message.
   - Click **6** to change the playback order of your messages (last-in/first-out or first-in/first-out).
   - Click **7** to set your Voice Mail Notification number. Follow the voice prompts. *For more information, see "Voice Mail Notification on a Digital Pager," page 194.*
   - Click **8** to set Dial by Name options. For more information, see *"Setting Up Your Name in the Dial by Name Directory," page 191*.
3. Click * to exit the **Setup Options** menu.

Creating or Changing Your Voice Mail Passcode

Creating a passcode for your Voice Mail is recommended to ensure security.

1. Gain access to your Voice Mail.
   - If prompted, click your passcode.
2. Click **2** to gain access to the **Setup Options** menu.
3. Click **3** to create or change your passcode.
   - You are prompted to enter a passcode. After clicking a new passcode, a voice announcement repeats the numbers you selected.

Recording a Voice Mail Greeting

A Voice Mail greeting is a recording of your voice that callers hear before they record a message in your Voice Mail.

1. Gain access to your Voice Mail.

   - If prompted, click your passcode.

2. Click **2** to gain access to the **Setup Options** menu.

3. Click **1** to record a Voice Mail greeting.

   - Click # when finished recording.

   - Click **2** to play back your greeting. If you need to make changes, repeat Step 3.

Setting Up Your Name in the Dial By Name Directory

A caller can reach you by entering the first three letters of your last name in your company's Dial by Name directory. You can also set up the system to reach you by entering the first three letters of your first name.

1. Gain access to your Voice Mail.

   - If prompted, click your passcode.

2. Click **2** to gain access to the **Setup Options** menu.

3. Click **8**, and then choose from three options:

   - Click **1** to make a voice recording of your name. Click **#** to stop recording. You are prompted to click the first three letters of your last name on the Keypad.

   - Click **2** to play back your recording.

   - Click **3** to remove your name from the Dial by Name directory.

Sending a Message to an Individual or a Work Group

At some point, you may need to send a Voice Mail message to an individual or a work group.

Sending a Message to an Individual

1. Gain access to your Voice Mail.

   - If prompted, click your passcode.

2. Click **7**.

3. On the Keypad click the extension of the individual receiving the message.

Note If you don't know the extension of the person you are calling, click 8 and you are prompted to click the first three letters of the person's last name to use the Dial by Name feature.

4. Click #.

 - Follow the voice prompts to record your message.

Sending a Message to a Range of Extensions

1. Gain access to your Voice Mail.

 - If prompted, click your passcode.

2. Click **7**.

For example, to dial the range of extensions 100 to 125:

1. On the Keypad, click the first extension in the range—**100**.
2. Click *****.
3. Click the last extension in the range—**125**.
4. Click #.
5. Follow the voice prompts to record your message.

Sending a Message to a Work Group

1. Gain access to your Voice Mail.

 - If prompted, click your passcode.

2. Click **7**.
3. On the Keypad, click the work group extension.
4. Click #.

 - Follow the voice prompts to record your message.

Forwarding a Voice Mail Message to an Individual or a Work Group

At some point, you may need to forward a Voice Mail message you received to another extension or group of extensions.

Forwarding a Voice Mail Message to an Individual

1. Click **7** after listening to a Voice Mail message.
2. On the Keypad, click the individual extension of the person receiving the message.

Note If you don't know the extension of the person you are calling, click 8 and you are prompted to enter the first three letters of the person's last name to use the Dial by Name feature.

3. Click **#**.

   - You are prompted to record a prefacing message. Use this time to give any necessary information to the person receiving the message.

Forwarding a Voice Mail Message to a Range of Extensions

1. Click **7** after listening to a Voice Mail message.

For example, to dial the range of extensions 200 to 225:

2. On the Keypad, click the first extension in the range—**200**.

3. Click *.

4. Click the last extension in the range—**225**.

5. Click **#**.

   - You are prompted to record a prefacing message. Use this time to give any necessary information to those receiving the message.

Forwarding a Voice Mail Message to a Work Group

1. Click **7** after listening to a Voice Mail message.

2. On the Keypad, click the work group extension.

3. Click **#**.

   - You are prompted to record a prefacing message. Use this time to give any necessary information to those receiving the message.

Gaining Access to Your Voice Mail From Another Internal Telecor VS1 Station

1. Dial **6 + your extension** from the station.

   - Wait for your greeting.

2. Click **#** when your greeting begins.

   - If prompted, click your passcode.

3. Follow the voice prompts accordingly.

Gaining Access to Your Voice Mail From an External Line

1. Dial your company telephone number.

   - If a receptionist answers, ask to be transferred to your Voice Mail.
   - If an Auto Attendant answers, dial **6 + your extension** when the Auto Attendant prompts you.

2. Click # when you hear your greeting.

   - If prompted, dial your passcode.

3. Follow the voice prompts accordingly.

Voice Mail Notification on a Digital Pager

The VS1 phone system can be customized to place a call to your pager when you receive a new Voice Mail message. This is an important feature when you are expecting a call, or message, and you are not at your office. Voice Mail Notification on a Digital Pager is also a good feature for after-hours emergencies or technical assistance.

In addition to sending a preassigned numeric message alerting you to new Voice Mail, Caller ID can be sent with the page so you can see who is leaving the message. You can even choose the number of times the page is repeated, and how often, until you check your new messages. You can customize your Voice Mail to alert up to three different pagers when you receive a new message.

It is also possible to redirect your new Voice Mail to up to 20 other extensions. Only new messages are redirected; old messages are left untouched.

There are two levels of settings for Voice Mail Notification: Basic Voice Mail Notification and Advanced Voice Mail Notification. Basic Voice Mail Notification is used to send a notice to a single pager. The Advanced settings are used to send a notice to multiple pagers, and redirect your new messages to other extensions.

Basic Voice Mail Notification

Basic Voice Mail Notification sends a notice to a single pager when you receive a new Voice Mail message. Basic Voice Mail Notification sends three paging attempts, with 15 minute intervals between attempts. These settings cannot be changed at the Basic level.

Gaining Access to Basic Voice Mail Notification Settings

1. Gain access to your Voice Mail.

   - If prompted, dial your passcode.

2. Click **2** to gain access to the Voice Mail Setup Options menu, and then click **7** to gain access to Voice Mail Notification.

3. Click **1** to set up Basic Voice Mail Notification.

Turning On Basic Voice Mail Notification on a Digital Pager

1. Enter the pager number for Voice Mail Notification followed by the # key. You do not need to enter a 9 prior to the pager telephone number.

2. Enter your PIN number followed by the # key. If your pager does not require a PIN number, press the # key only.

3. Enter the numeric message you want displayed on your pager followed by the # key. You can enter up to 24 digits.

4. To turn on Caller ID display, click **1**, or to turn off Caller ID display, click **2**.

Note Caller ID must be installed on your phone system for this feature to be active. See your system installer.

- You hear a playback of all the numbers you have entered—pager number, PIN number (if needed), numeric message—and if Caller ID is turned on or off. If any of these numbers or settings is incorrect, you can change them by repeating Steps 1-4.

Turning Off Basic Voice Mail Notification on a Digital Pager

1. Gain access to the Basic Voice Mail Notification Settings by following Steps 1-4 of "Gaining Access to Basic Voice Mail Notification Settings."

2. Click only the # key when prompted to enter a pager number for Voice Mail Notification. This clears the pager number entered, if any, and no pages are sent upon receipt of new Voice Mail.

Note To turn Basic Voice Mail Notification on a Digital Pager back on, you must repeat Steps 1-4 of *"Turning On Basic Voice Mail Notification on a Digital Pager"* above.

Advanced Voice Mail Notification

Advanced Voice Mail Notification provides several options for sending pager notification upon receipt of a new Voice Mail message. Through the Advanced options you can customize your Voice Mail to contact up to three pagers, set the number of times pages are sent until answered, and set the time interval between pages. The Advanced level also gives you the option to redirect your new Voice Mail messages to other extensions in your office.

Gaining Access to the Advanced Voice Mail Notification Options

1. Gain access to your Voice Mail.

   - If prompted, dial your passcode.

2. Click **2** to gain access to the Voice Mail Setup Options menu, then click **7** to gain access to Voice Mail Notification.

3. Press **2** to set up the Advanced Voice Mail Notification options.

Setting Up Voice Mail Notification to up to Three Digital Pagers

1. Click **1** while in the Advanced Voice Mail Notification options of your Voice Mail. This enables you to enter the settings for the first pager.

2. Enter the pager number for Voice Mail Notification followed by the # key. You do not need to enter a 9 prior to the pager telephone number.

3. Enter your PIN number followed by the # key. If your pager does not require a PIN number, click the # key only.

4. Enter the numeric message you want displayed on your pager followed by the # key. You can enter up to 24 digits.

   - You hear a playback of all the numbers you have entered—pager number, PIN number (if needed), numeric message. If any of these numbers or settings is incorrect, you can change them by repeating Steps 1-4.

5. Repeat Steps 1-4 to enter the settings for a second and third pager. In Step 1, click **2** or **3**, respectively, for the second or third pager.

Canceling Voice Mail Notification to Multiple Pagers

1. Click **1** while in the Advanced Voice Mail Notification options of your Voice Mail. This enables you to enter the settings for the first pager.

2. Click only the # key when prompted to enter a pager number for Voice Mail Notification. This clears the pager number entered, if any, and no pages are sent.

3. Repeat Steps 1-2 for the second and third pagers if you are canceling notification to those as well, click **2** or **3** respectively, for the second or third pager.

Note If you are canceling notification to only one of the pagers, change only that pager's settings. Notification to the other pagers remains as customized.

Setting the Number of Paging Attempts

1. Click **5** while in the Advanced Voice Mail Notification options of your Voice Mail.

2. Enter the maximum number of times to be paged, followed by the # key.

Note 10 is the highest number of attempts that can be set. If you enter a value greater than 10, the system sets the attempts to 10.

Setting the Paging Interval

1. Click **6** while in the Advanced Voice Mail Notification options of your Voice Mail.

2. Enter the number of minutes between pages, followed by the # key.

Note The Paging Interval can be set from a minimum of 1 minute to a maximum of 34,464 minutes. If you enter an interval greater than 34,464 minutes, the system sets the interval to 34,464 minutes.

Redirecting New Voice Mail Messages to Other Extensions

1. Click **7** while in the Advanced Voice Mail Notification options of your voice mail.

2. Enter the list of extensions for Voice Mail redirect. After you enter an extension, you hear a voice recording verifying the extension entered.

Note If you do not know the extension of a person, click 8 to enter a Dial by Name directory. This directory allows you to enter the first three letters of a person's last name, instead of their extension.

- You can enter up to 20 extensions.

3. Click the **#** key when your extensions list is complete.

Turning Off Redirecting New Voice Mail Message

1. Click **7** while in the Advanced Voice Mail Notification options of your Voice Mail.
2. Click only the **#** key when prompted to enter the list of extensions for Voice Mail redirect. This clears any extensions set up to receive your redirected Voice Mail messages.

Reference

OVERVIEW OF REFERENCE

This section provides additional information not covered in the previous sections. This information is provided as reference, and does not describe in detail the configuration of the system or hardware, which is found in the appropriate sections of this guide.

Included in this section are:

- Default Extensions and Access Codes
- An overview of Direct Inward Station Access (DISA) and use
- An overview of Digital Signal Processor (DSP) Volume Adjustments
- A procedure for setting the system date and time from a phone station
- Flow Charts of system processes for handling calls, transfers, ACDs and Auto Attendants
- An overview of how to interpret Line Status symbols when the **`ls`** command is issued
- A reference to PBX commands
- Recording Voice Files
- Pre-recorded Message List and Phrase List
- A procedure for Scheduled Disk Optimization
- An overview of Station Message Detail Recording (SMDR) and how to interpret the data
- An overview of the VS Call Accounting application to view SMDR data.
- A procedure for Verifying CO Disconnect Signals
- Setting up a T1 Interface Card
- A glossary of common telephony and computer telephony features and terms, and how they relate to the VS1 telephone system

DEFAULT EXTENSIONS & ACCESS CODES

Default Extensions

The VS1 telephone system comes installed with a number of preconfigured extensions. Take note when programming extension numbers to stations.

1st digit	# of digits	# Range	Use
0	1	0	operator
1	3	100–109	DP200 Stations
3	3	300-319	Guest Mail Boxes
4	3	421–430	ACDs
		431–433	Conference Rooms
		441–460	Paging Zones
5	3	599	Reserved for RSA modem as extension

Access Codes

The following access codes are used by the VS1 system.

1st digit	# of digits	# Range	Use
6	4	6+Extension	Voice Mail
7	4	7101–7199	Command Files **`f1.cmd`**–**`f99.cmd`**
		7238–7257	Auto Attendants (7238 accesses AA1, 7239 accesses AA2, etc.)
		7800	Date/Time
		7801	Function Key Programmer
8	2	80, 81, 82	Outside Lines
9	varies	9	Outside Line

DIRECT INWARD SYSTEM ACCESS (DISA)

Direct Inward System Access (DISA) is a way to gain access to the internal dial tone of the VS1 Telephone System from an external phone not connected to the system. The VS1 System only allows internal dial tone from station Extension ports, and DISA allows you to call into a CO port on the system to get internal dial tone. DISA allows features such as:

- Running Command Scripts
- Dialing internal extensions
- Placing external calls
- Dialing extensions of Paging Zones
- Dialing extensions of Auto Attendants
- Dialing extensions of ACDs

The main steps to setting up DISA on the VS1 System include:

1. Enabling DISA on at least one CO port
2. Targeting a CO port for DISA.
3. Setting up DISA passwords

DISA Example

1. From an outside telephone, dial a CO port telephone number configured for DISA operation.

2. After hearing the first ring, but before or during the second ring, press #.

   - The CO line ceases ringing and no tone is heard. The silent line is a security measure designed to prevent unauthorized access to the VS1 phone system.

3. Dial the extension number followed by the DISA password. If the correct code is entered, three beeps sound and the caller hears the system internal dial tone. If an incorrect password is entered, the line immediately disconnects.

4. Use * to return to internal dial tone anytime ringing is heard during DISA operation

Enabling DISA on a CO Port

Many VS1 System installers choose to make DISA available on only one CO port so that it is not generally accessible to every end-user at a particular site. Providing access to a limited number of people can help reduce fraud and abuse of the DISA feature.

Note	It is more effective to use a CO port that is not in a hunt group so that the CO number is always available to the caller wanting DISA access. If the CO port is part of a hunt group, it should be in the middle of the hunt group.

1. In the VS1 Editor Tree Control display, click **Port Configurations**.

2. In the **Port Configurations** pane, select the configuration where you want to make changes. For example, select the DAY configuration.

3. Select the CO port that DISA will be enabled on.

4. Under **Rings**, set the CO port to ring the first target for two rings.

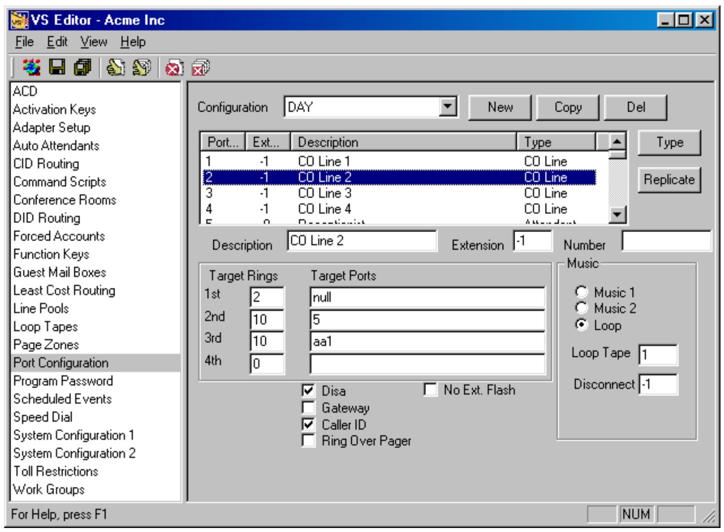
5. Under **Target Ports**, set the target by typing **null**. The **null** device doesn't ring a physical extension. When Target Ports is set to **null**, the call rings nowhere for however many rings set up under Rings. Generally the rings are set to no more than two. This allows the system to listen for the extension and passcode for DISA operation.

6. Set the number of rings for the remaining targets according to your configuration.

7. Select the **DISA** check box.

8. Make sure you repeat the changes made to this port in all configurations, not just for the DAY configuration.

DISA setup on CO Port



Setting up a DISA Password on an Extension Port

1. In the VS1 Editor Tree Control display, click **Port Configurations**.

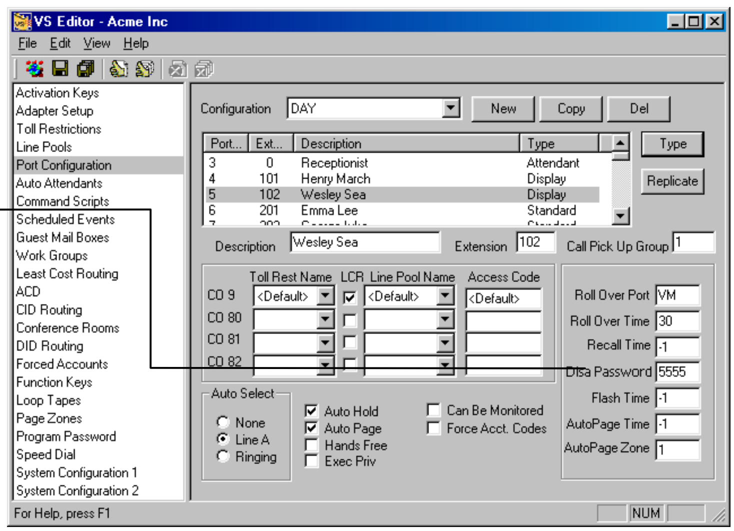
2. In the **Port Configurations** pane, select the configuration where you want to make changes. For example, select the DAY configuration.

3. In the **Port Configurations** pane, select the Extension port of a user you want to give a DISA account password.

4. In the **DISA Password** text box, enter a five-digit number ranging from 00000–99999. By default, the DISA Password is set to –1, which means no account has been assigned, and no DISA features are available for that extension.

5. Make sure you repeat the changes made to this port in all configurations, not just for the DAY configuration.

DISA Password text box



DISA password on extension port

DSP VOLUME ADJUSTMENTS

The VS1 telephone system enables CO and Extension volume adjustments through the Host Adapter Digital Signal Processor (DSP). Although the DSP has been programmed to values that should work in most settings, there are situations (such as trunk-to-trunk transfers) that may require different values. DSP volume adjustments are most commonly made on CO ports that handle transfers to outside lines.

Each port (T1 or Port Expansion Unit) has two DSP settings associated with it. One setting controls the volume level into the port (IN), and the other setting controls the volume setting out of the port (OUT). (T1 channels do not require volume adjustments.)

The volume level of a connection through the Telecor Voice Server (TVS) is determined by adding the volume level of each component of the connection. For example, the level heard by the extension party on an outside CO call is the sum of the CO IN volume and the Extension Port OUT volume. Likewise, the level heard by the outside CO party is the sum of the Extension Port IN volume and the CO OUT volume.

Note These settings do not affect internal calls. Volume adjustments for CO calls to Auto Attendants or Voice Mail use the set `aavol[in][out]= [n]` command.

The command syntax associated with DSP volume is:

`dsp vol {co|ext|`*`port`*`} ={`*`n`*`} {in|out}` (Where n = -12 to 12).

Sample DSP Volume Adjustments

`dsp vol co =5 in`	Set all incoming CO volumes to 5.
`dsp vol ext =-2 out`	Set all outgoing extension volumes to -2.
`dsp vol 37 =4 in`	Set port 37 volume IN to 4.
`dsp vol co 0 in`	Show the current CO IN settings.

Default Values	Analog CO Port	T1 CO Port	Extension Port
Volume IN	0	0	0
Volume OUT	0	0	0

Note The **`dsp enable`** command enables volume adjustments and must be issued before any **`dsp vol`** commands are recognized by the system.

DSP volume adjustments remain effective until the system is reset. *To make permanent changes, see "Adjusting DSP Volume through the Startup Command Script" below.*

When making DSP adjustments, keep in mind that changing any one setting has an effect on other settings. Things to remember:

- Increasing the Volume OUT to an extension also increases the sidetone returned from that extension.
- Setting the Volume OUT on a CO to greater than 1 can create outbound dialing problems on the system.
- Lowering the Volume IN from an extension reduces the background noise picked up by that extension.
- The Volume IN from a CO controls the level at which Voice Mail is recorded on the system.

Adjusting DSP Volume through the Startup Command Script

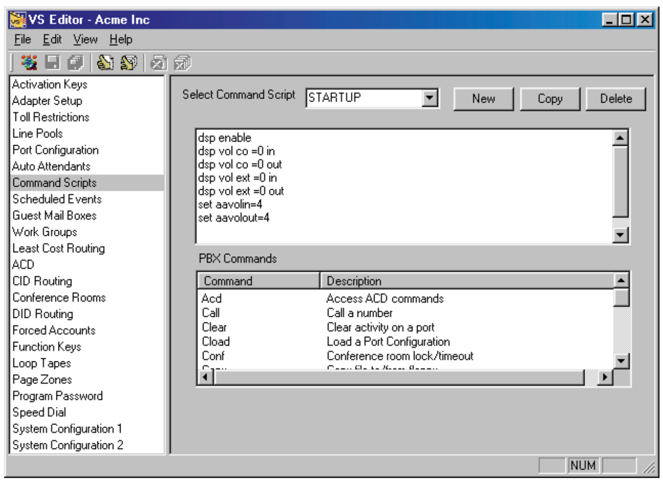
By default, the DSP Volume IN and OUT for CO and Extension ports are set to the values in the **Startup** Command Script. The **Startup** Command Script is run every time the system boots up. Therefore, the DSP volume adjustments are made every time the system starts.

WARNING

Warning! Use caution when adjusting the DSP volume settings. Improper DSP volume settings can result in poor sound quality, feedback, or telephone company fines.

To adjust these values complete the following steps:

1. In the VS1 Editor application, select **Command Scripts**.
2. Select **STARTUP** in the **Select Command Script** dropdown box.
3. In the Startup Command script window, adjust the DSP Volumes to the new values.



Command Scripts window

4. Click the **Save** button in the toolbar.

Adjusting DSP Volume on CO Line Connected to an Auto Attendant or Voice Mail

To adjust the DSP volume on a CO line that is targeted to an Auto Attendant or Voice Mail, a different command syntax is used. The volume adjustments have a wider range of values than a CO or Extension port as previously described.

Note The **`dsp enable`** command enables volume adjustments and must be issued before any **`aavolin`** or **`aavolout`** commands are recognized by the system.

Use the following commands to adjust the DSP volume on a CO Port configured to an Auto Attendant or Voice Mail:

DSP Volume IN

`set aavolin={n}` where n = -12 to 12 (The default setting is 0). For example: **`set aavolin=3.`** This value is in addition to any change made in the **`dsp vol co={n} in`** command.

DSP Volume OUT

`set aavolout={`*`n`*`}` where n = -12 to 12 (The default setting is 0). For example: **`set aavolout=3.`** This value is in addition to any change made in the **`dsp vol co={`*`n`*`} out`** command.

SETTING THE SYSTEM DATE AND TIME

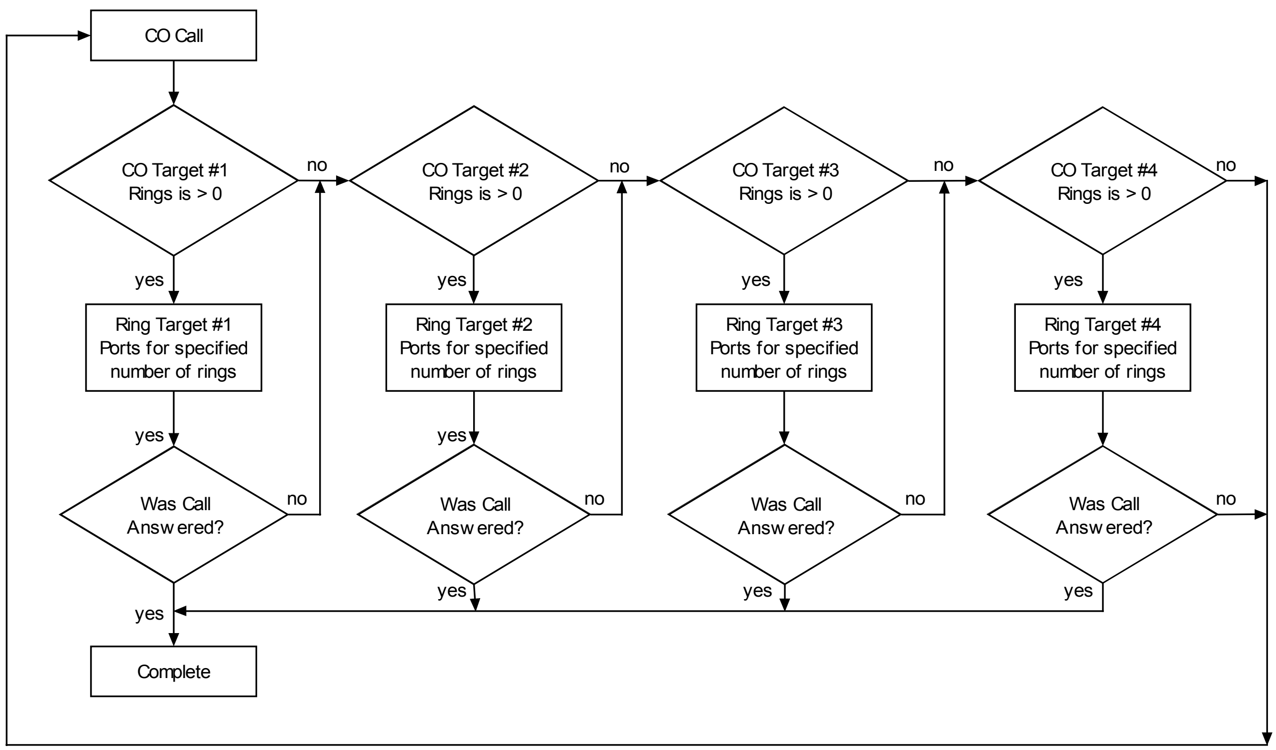
The date and time of the VS1 phone system can be set from any phone. To set the date and time, complete the following steps:

1. Lift the handset, and then dial **7800**.
2. Press **1** to hear the current date and time.
3. Press **2** to change the date and time.
4. Use the Keypad to enter the month, day, year, hour and minutes. Use military time. For example enter 7:45 PM on October 18, 2003 as 1018031945.
5. Hang up.
6. Verify the date and time on a Display Phone Model 200 (DP200).

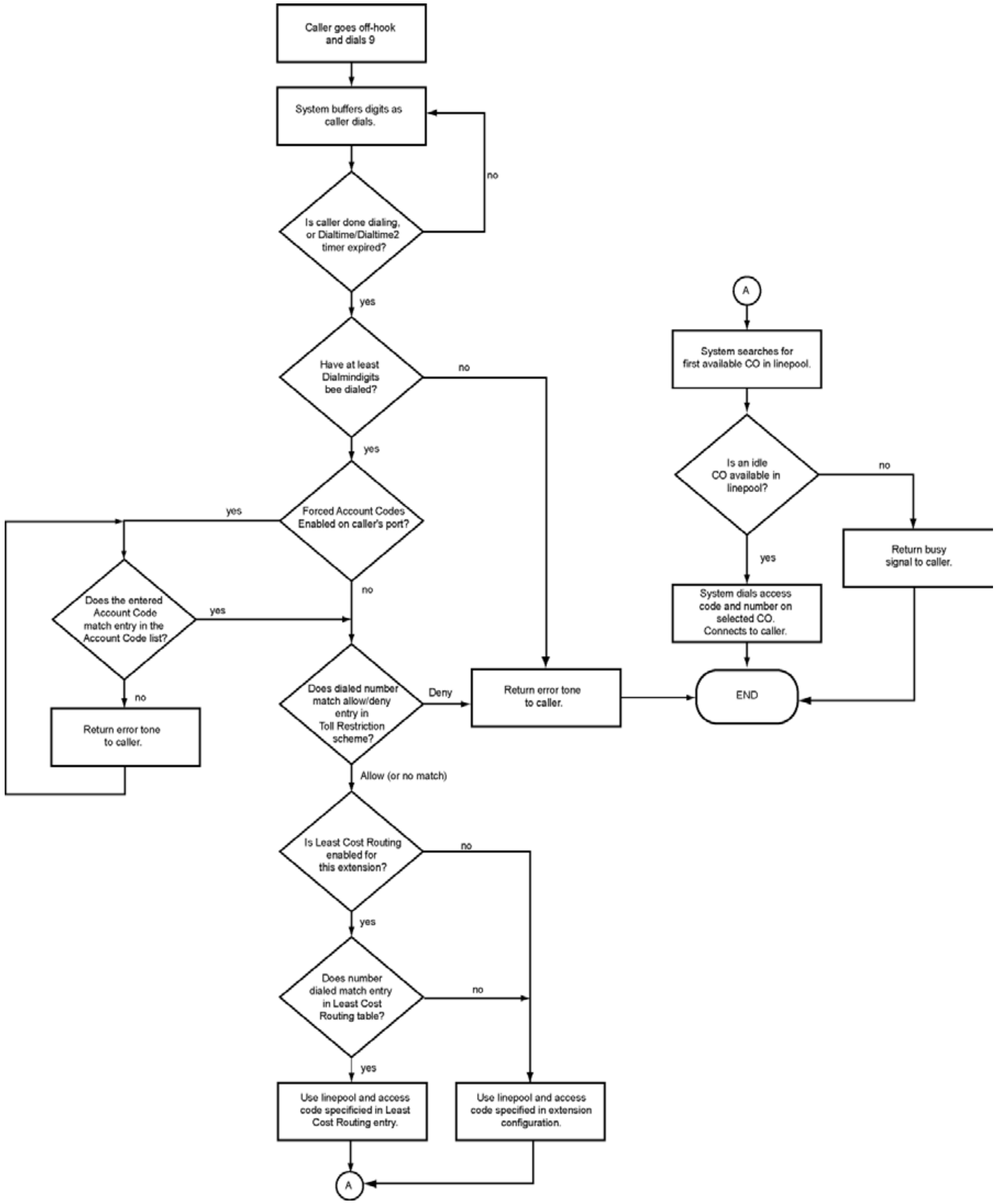
TELECOR VOICE SERVER FLOW CHARTS

This part of the "Reference" section provides a series of flow charts, outlining the procedures followed by the Telecor Voice Server for handling inbound and outbound calls, and transfers. Also charted is how calls to Auto Attendants and ACDs are processed. These charts are to be used in conjunction with the detailed descriptions found in the *"VS1 Editor" section of this guide.*

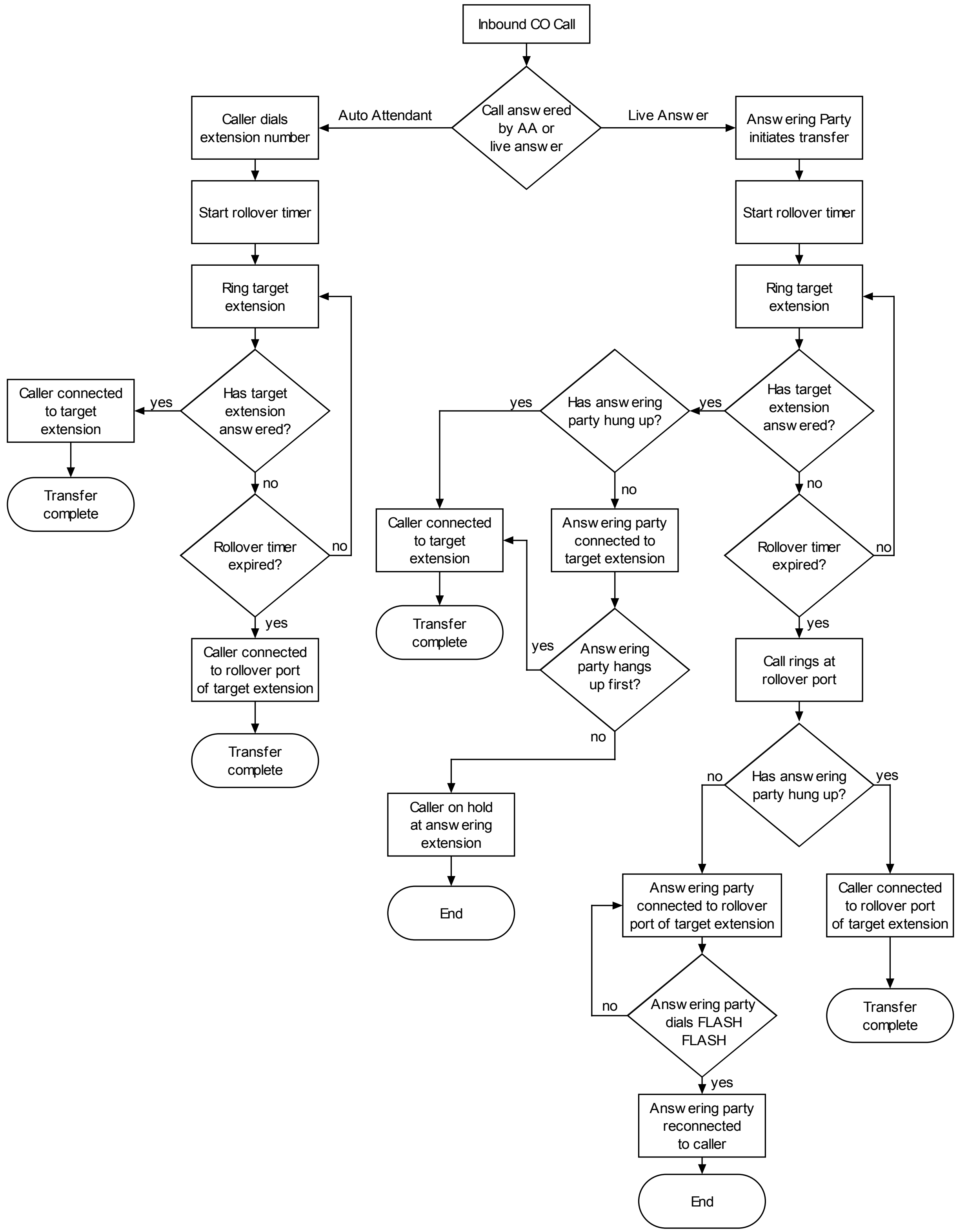
Inbound CO Call



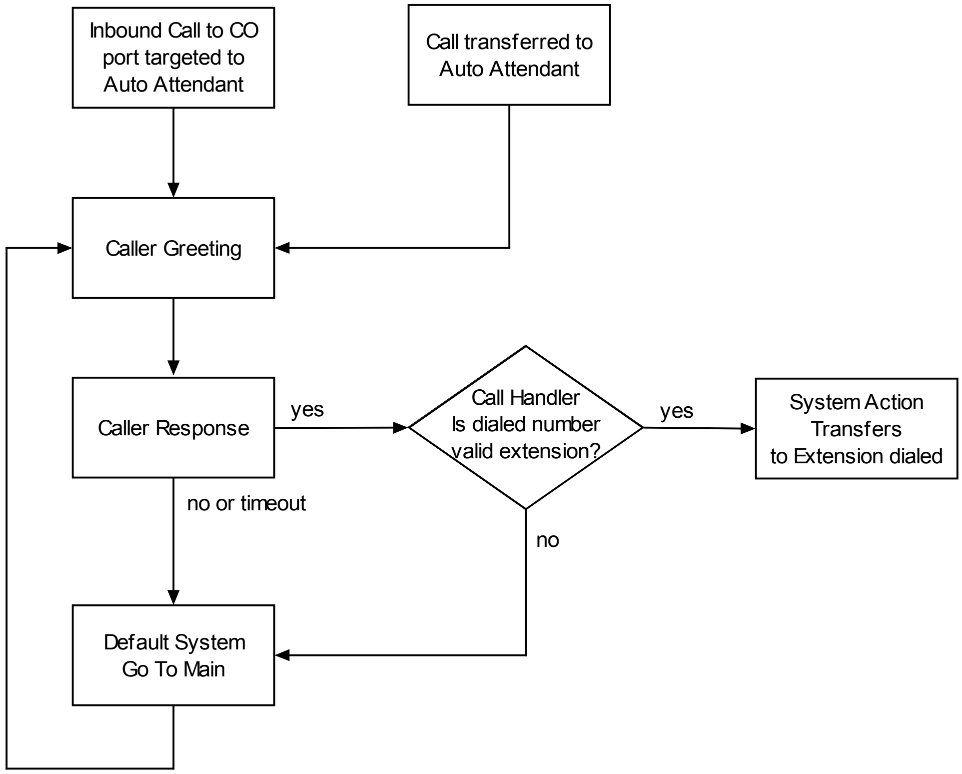
Outbound CO Call



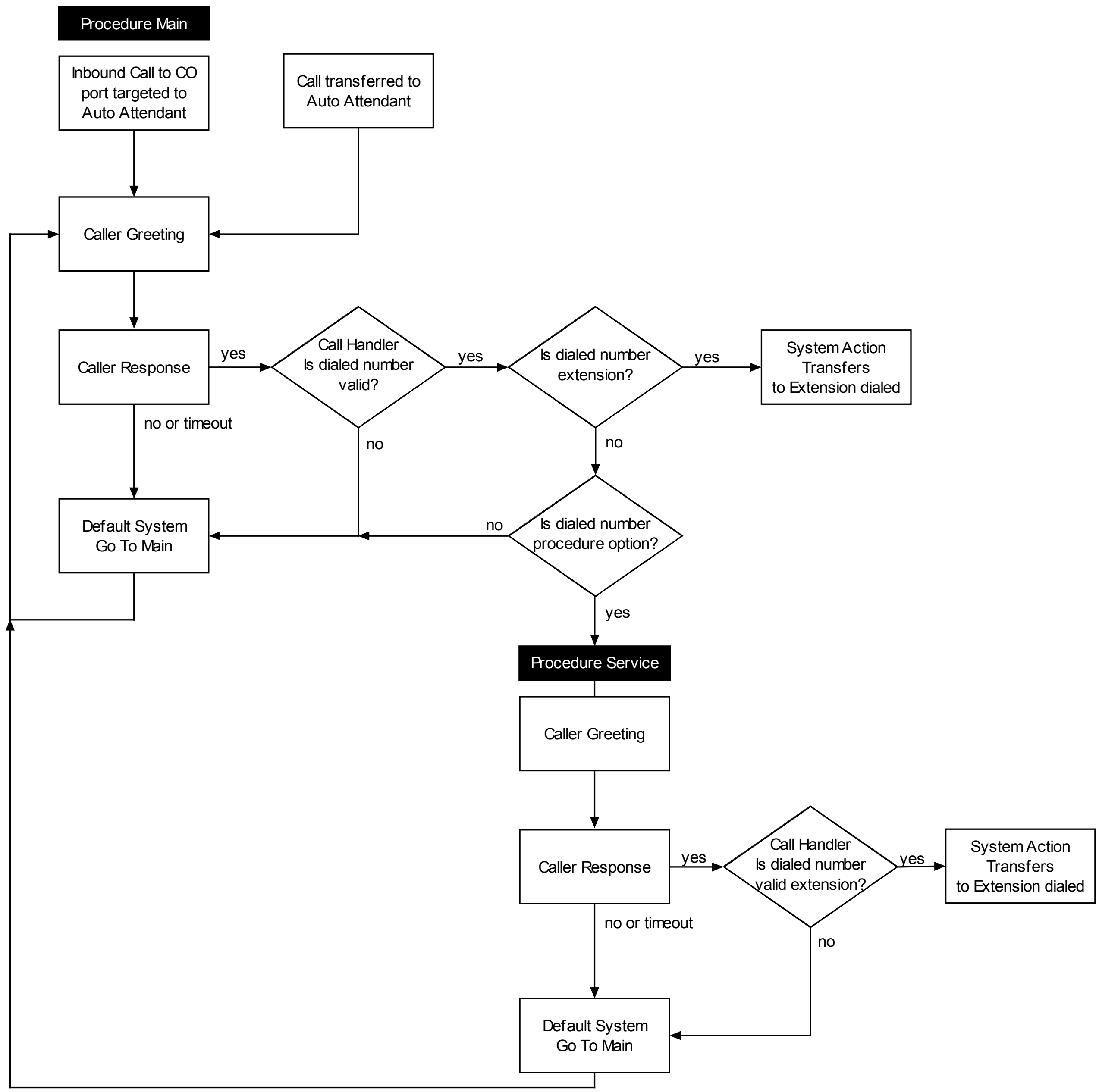
Call Transfer



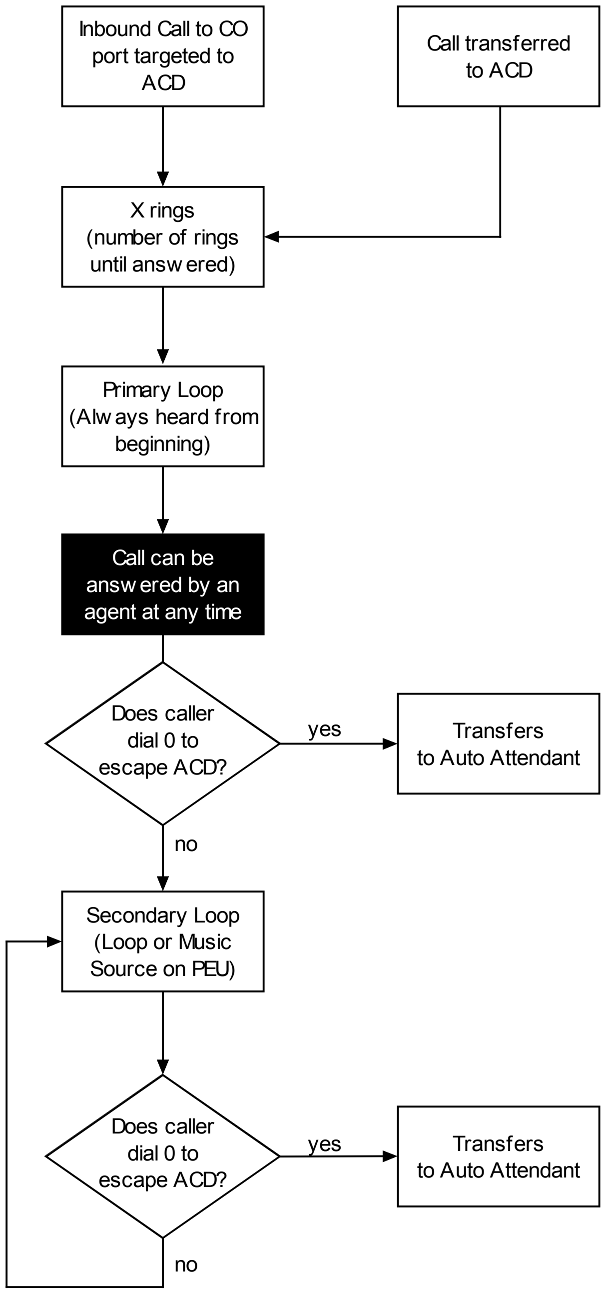
Auto Attendant—Single Procedure



Auto Attendant—Multiple Procedure



Automatic Call Distributor



LINE STATUS SYMBOLS

The following is a description of the symbols displayed when the **`ls`** command is entered at the Telecor Voice Server (TVS) Command prompt. The **`ls`** command provides a display of the Line Status of each port on the VS1 telephone system.

```
Port:Ext   Desc                          Stat
   1: ---  CO Line 1                     C-Idle
   2X ---  CO Line 2                     C-Idle
   3: ---  CO Line 3                     C-Idle
   4: ---  CO Line 4                     C-Idle
   5:    0 Receptionist                  -
   6:  100 Norm Stenger                  ___ ...
   7%  101 Walter O'Riley                ___ ...
   8:  599 Board Room                    ___ ...
   9:  103 M Kensington                  ___ ...
  10:  104 Cliff Fitzgerald              s_{1}___ ...
  11%  105 James Washington              ___ ...
  12:  106 George Banks                  s_{1}___ ...
  13:  107 Johanna Hartman               s_{1}___ ...
  14:  108 Matt Sanderson                s_{1}___ ...
  15:  109 Jenny Farley                  s_[A]___ ...
  16:  110 Autie Fleishman               s_{1}___ ...
  34: ---  T1 1                          D-Idle
  98:  102 T1 1                          C-Idle
  99: ---  T1 2                          C-Idle
 100: ---  T1 3                          C-Idle
 116: ---  T1 15                         C-Idle
 128: ---  T1 24                         C-Idle
 224: ---  Voice Mgr 1
```

Line Status window

Port: column: Displays the port number on the Port Expansion Unit(s).

`%` - port is on Do Not Disturb

`X` - port is disabled

Ext column: Displays the Extension number assigned for current configuration period

`---` - port is CO, T1 or Voice Mgr

Desc column: Displays the description entered for the port.

Stat column: Displays the status of the port, based on the type of port assigned. Where the port type format includes an x and y, for example xy[z], the x refers to the speaker statuses and the y refers to the ACD statuses.

Standard Phone-Type 1 (Format xy[z])
No status symbols

Telecor Display Phone-Type 7 (Format xy{#})

`s` - speaker off

`S` - speaker on (speakerphone or paging)

`_` - not using ACD

`a` - ACD logged on and in wrap-up

`A` - ACD logged on

`#` - number indicates active line

Modem/Fax-Type 6
No status symbols

Telecor Attendant-Type 8

+	-off-hook
–	-on-hook
	If no status, Activation Key not entered or invalid

Telecor Connect-Type 9

+	-off-hook
--	on-hook
dialing	line cord is unplugged

CO Port-Type 2

C	-normal
D	-DISA enabled
-Idle	-CO line is not being used
-Wait-CID z	-PBX is waiting for Telco to send CID information
--> Description z	-CO line is connected to Description port
-Ring: Group n z	-PBX is ringing the nth target port(s)
W-> Description z	-CO target port has answered; call ringing another extension
d-> Extension (0)	-PBX is dialing on this CO
-Alerted	-PBX is searching for CO to make outbound call
H-> Description z	-Call is on hold

Press the spacebar to open the Polling window in the Line Status window. The first column represents a device polled and responding correctly; the second column represents a device polled with no response; the third column represents a device polled with an invalid response. To restart the counters, type r.

```
Port:Ext   Desc                           Stat
   1: ---  CO Line 1                             0     50473        0
   2X ---  CO Line 2                             0         0        0
   3: ---  CO Line 3                             0         0        0
   4: ---  CO Line 4                             0         0        0
   5:    0 Receptionist                   ---
   6:  100 Norm Stenger                          0         0        0
   7%  101 Walter O'Riley                        0         0        0
   8:  599 Board Room                            0         0        0
   9:  103 M Kensington                          0         0        0
  10:  104 Cliff Fitzgerald                7004884         1        0
  11%  105 James Washington                      0         0        0
  12:  106 George Banks                    7004886         0        0
  13:  107 Johanna Hartman                 7004898         0        0
  14:  108 Matt Sanderson                  7004894         0        0
  15:  109 Jenny Farley                          0    225965        0
  16:  110 Autie Fleishman                 7004889         1        0
  34: ---  T1 1                                  0         0        0
  98:  102 T1 1                                  0         0        0
  99: ---  T1 2                                  0         0        0
 100: ---  T1 3                                  0         0        0
 116: ---  T1 15                                 0         0        0
 128: ---  T1 24                                 0         0        0
 224: ---  Voice Mgr 1                    ---
```

Polling window

TELECOR VOICE SERVER SYSTEM COMMANDS

The following commands can be issued from the command line of the Telecor Voice Server, or from the **Terminal** window of the Tel-Site system management application. Each command can be used to monitor a specific aspect of the system, and can be issued while the system is running, having no impact on the system functionality.

argument	Indicates an argument for which you must supply a value.
{x}	An argument or a constant within braces { } is required.
[x]	An argument or a constant within square brackets [ ] is optional.
x|y|z	Constants or arguments separated by a vertical bars requires a choice.

Command Syntax	Result
`exit`	**Shuts down the TVS software and reboots the TVS (from local console only)**
`call ext {`*extension*`} {`*sysfile*`}`	**Dials extension and plays voice file**
`call pager {`*sysfile*`} {`*pagezone*`}`	**Plays voice file over specified paging zone**
`call remote {`*number*`} {`*sysfile*`} [`*dtmf*`]`	**Access CO line, dials number, plays file, sends DTMF**
`clear {`*port*`}`	**Clears all activity on a port and turns off Message Waiting Lamp**
`cload {`*file*`}`	**Loads new configuration**
`conf lock 1`	**Enables conference room to be locked**
`conf lock 0`	**Disables conference room locking**
`conf timeout [`*n*`]`	**Determines how long (minutes) conference including a CO is allowed to run**
`conf timeout 0`	**Disables conference timer**
`copy {`*source*`} {`*destination*`}`	**Copies file**
`date [`*mmddyyhhmm*`]`	**Sets month, day, year, hour and minutes (Military time) i.e. 0521981550**
`del {`*filename*`}`	**Deletes specified file**
`dir {`*filename*`}`	**Displays list of files in specified path**
`disable {`*port*`|all}`	**Prevents users from initiating call on specified CO port**
`dsp enable`	**Enables adjustments to be made to DSP settings**
`dsp vol {co|ext|`*port*`} ={`*n*`} {in|out}`	**Sets DSP volume for specified port**
`dump call`	**Displays PBX call buffer**
`dump page`	**Displays PBX page buffer**
`enable {`*port*`|all}`	**Cancels the disable command**
`err`	**Displays status of DSP, Temperature and Line Card**
`fwd`	**Displays list of forwarded ports**
`grpid 1`	**Starts Group ID**
`grpid 0`	**Stops Group ID**
`help`	**Displays listing of PBX commands**
`ls`	**Displays the line status of all extensions and CO ports**
`ls acd`	**Displays status of all ACDs**
`ls co`	**Displays status of CO ports**
`ls ext`	**Displays status of Extension ports**
`modem answer`	**Causes modem to answer**
`modem reset 1`	**Resets the modem**

`modem send {command}` .... **Sends AT commands to be sent to the modem**
`pause {n}` .... **Pauses a command script for amount of time (seconds)**
`relay {1|2} {time}` .... **Turn on designated relay for specified time (hundredths of seconds)**
`ren {filename} {newfilename}` .... **Renames file**
`rop 1` .... **Causes incoming calls to ring over pager outputs**
`rop 0` .... **Turns off ring over pager command**
`set` .... **Shows current system settings**
`set aavolin={n}` .... **Adjusts inbound volume for COs connected to AA or VM to specified level**
`set aavolout={n}` .... **Adjusts outbound volume for COs connected to AA or VM to specified level**
`set animode={0|1}` .... **Reserved. Do not change.**
`set autopagerepeat={n}` .... **Sets time interval (seconds) before AutoPage repeats**
`set autopagetime={n}` .... **Sets time (seconds) of the initial AutoPage**
`set coholdtime={n}` .... **Sets system default time (seconds) for parked and held calls before ringback**
`set dialmindigits={n}` .... **Determines minimum number of digits that must be dialed**
`set dialtime1={n}` .... **Sets the time (seconds) after each dialed digit before assuming dialstring complete**
`set dialtime2={n}` .... **Sets the time (seconds) the system will wait before assuming dialstring complete**
`set dialtimerdigits={n}` .... **Determines how many digits must be dialed before DialTime2 begins**
`set dndtime={n}` .... **Determines time (hours) for DND on standard phones**
`set dndtime=0` .... **Turns DND timer off for standard phones**
`set loop{n}={file}` .... **Specifies voice file associated with a loop tape**
`set ring0={n}` .... **Determines length (hundredths of seconds) of ring to be valid**
`set ring1={n}` .... **Determines length (hundredths of seconds) of time before waiting for next ring**
`set ring2={n}` .... **Determines length (hundredths of seconds) of time before recognizing next valid ring**
`set smdrmode27={0|1}` .... **Changes number of year digits in SMDR Date fields.**
`show acd {acd number}` .... **Displays status for specified ACD**
`show events` .... **Displays remaining scheduled events for week**
`show ports {file}` .... **Displays port configuration information**
`shutdown {n}` .... **Resets system in specified time (minutes)**
`shutdown idle` .... **Turns the system off**
`stat` .... **Displays Process Status**
`stop` .... **Stops operating system**
`sysinfo` .... **Displays system information**
`time` .... **Displays system time**
`t1 comm {0|1}` .... **Displays version and communication stats about T1 card 1 (0) or 2 (1)**
`t1 setup {0|1}` .... **Displays T1 settings for T1 card 1 (0) or 2 (1)**
`t1 stat 0|1` .... **Displays status of specified T1 card**
`vm {extension}` .... **Displays voice mail information for specified extension. Press enter to view additional stats**
`vmaint scan [forward to]` .... **Checks integrity of voice mail; forwards orphaned to specified extension or deletes**
`vmaint purge {extension|*} {daysold} [forward to]` .... **Purges voice mail to specified extension**

RECORDING VOICE FILES

Voice Files recorded for Auto Attendants and Loops Tapes are stored as files using Pulse Code Modulation (PCM) format. These files can be identified by the file extension **`.pcm`**. The file name begins with **`$va`** followed by a three-digit number such as, **`$va001.pcm`** or **`$va123.pcm`**. The three-digit number after **`$va`** is the number you set up for that voice file. Voice files can be recorded in one of two ways:

- Using the Voice Message Recorder
- Using the base system Port Expansion Unit Model 205 (PEU 205) or Dry Contact Unit (DCU) and the TVS Command Prompt.

For the best possible sound quality, Telecor recommends that you use an audio source connected to a PEU 205 or DCU.

Recording Using the Voice Message Recorder

To record using the Voice Message Recorder, complete the following steps:

1. Dial **7239** from any station option on the VS1 phone system. The following message is heard: "You have activated the Telecor System Voice Recorder. Please enter your passcode."
2. Dial **86423** (V-O-I-C-E). The following message is heard: "Thank you. To record, press 1. To play, press 2. To exit, press *."
3. Press **1** to record. The following message is heard: "Enter the number of the file you wish to record."
4. Enter a file number between one and three digits. For example, enter 100 for the file number.

Note: Files can be numbered 0-10 and 14-899. Telecor reserves prerecorded voice file numbers 11-13 and 900-999. *See page 222 for a list of those messages*. If a file number is already in use, the new voice file replaces the existing voice file.

5. Wait for the beep, and then speak into your handset or headset. Press **#** to end the recording. Record the message. Hang up or press ***** to end your recording session.

   For example, if you choose to record and enter 100, the voice file **`$va100.pcm`** is created and stored on the TVS in the **`c:\ops\va`** directory. To listen to any message you record, press **2** after Step 3, and then enter the number of the file you want to hear. If you enter a file number not recorded, you hear a two-beep error tone.

Recording Using the PEU-205 or DCU

An audio source, such as a tape recorder, connects to one of the two music inputs on the PEU-205 or DCU. The VS1 System accepts 600 ohm, low-impedance signals. A cable is required to connect the audio source to the terminals on the PEU-205 (or DCU). The PEU-205 and DCU accept spade connectors, number 6 stud, or bare wire. Use 24-gauge wire or larger. *See the Hardware section for connecting an audio source.*

1. Connect one end of the cables to the output jacks on the audio source. Next, connect the ground wire to the negative input terminal on the PEU-205 (or DCU) and the positive wire to the positive input terminal on the PEU-205 (or DCU).

2. Access the TVS Command Prompt via the Terminal window in Tel-Site.

3. At the TVS Command prompt, you must use the rec command and specify the Music 1 or Music 2 inputs. Assuming the voice file will be 500 and the audio source is connected to the first input, type the command in the following format: **`rec 1 va\$va500.pcm`**

Note: Files can be numbered 0-10 and 14-899. Telecor reserves prerecorded voice file numbers 11-13 and 900-999. *See page 222 for a list of those messages.* If a file number is already in use, the new voice file replaces the existing voice file.

- A display message states: "Acquiring DMA Channel... hit ESC to terminate. Press any key to record, ESC to cancel."

4. Start the tape recorder (or other audio source) and then press any key on your keyboard.
   - A display message states: "Press any key to stop recording." A series of dots appears to indicate that the system is recording.

5. When the message is finished, press ESC to stop the recording.

6. Press any key to return to the TVS Command prompt.

Playing a Voice File Using the Command Prompt

To play the voice file at an extension using the Command Prompt, complete the following steps:

1. Press the SPEAKERPHONE button or lift the handset on the telephone that will play the message.

2. At the TVS Command Prompt type **`call ext`** ***`{extension} {file}`***. For example, **`call ext 36 44`** calls extension **36** and plays voice file **$va044.pcm**.

3. Press ENTER.

PRE-RECORDED MESSAGE LIST

The following list shows the voice file number and the message it plays. you can use these messages when creating Auto Attendants.

011—“There are too many calls for the number of agents. Please add another agent.”
012—“Calls in queue have been waiting too long. Please add another agent.”
013—“Calls are in queue and zero agents are logged in. Please add agents.”
900—“You have activated the Telecor Voice Recorder.”
901—“To record press 1. To play press 2.”
902—“Enter the number of the file you wish to record.”
903—“Enter the number of the file you wish to play.”
904—“Thank you for calling.”
905—“You have reached the Telecor VS1 Phone System Main Menu.”
906—“If you know your party’s extension, you may enter it at any time.”
907—“To access a Voice Mail box, press 6.”
908—“For a directory, or to dial by name, press 555.”
909—“To reach the Operator, press 0.”
910—“If you are calling from a rotary phone, please stay on the line and I’ll transfer you to the Operator.”
911—“Please hold while I transfer you to...”
912—“The Operator.”
913—“Please hold while I transfer you.”
914—“I’m sorry, the directory has not been configured on this system. Dialing by name is not possible.”
915—“I’m sorry, the Operator is not available.”
916—“Please enter the first three digits of the person’s last name.”
917—“Please enter the first three digits of the person’s first name.”
918—“For Q or Z press 1.”
919—“For Q press 7, for Z press 9.”
920—“I’m sorry, that extension is busy.”
921—“If you would like to leave this person a message, please stay on the line.”
922—“Otherwise...”
923—“or...”
924—“To return to the Main Menu, press star.”
925—“Please hang up and try your call again later.”
926—“After leaving your message, you may press pound for further sending options, or simply hang up.”
927—“To exit, press star.”
928—“I’m sorry, that is not a valid option.”
929—“Invalid passcode.”
930—“Access denied.”
931—“New message.”
932—“New messages.”
933—“No new messages.”
934—“Message deleted.”
935—“Please re-enter your passcode.”
936—“The person you have called is not able to speak with you at this time. Please leave your name and number at the sound of the tone. Press the pound sign when finished for further options.”
937—“Please enter your new passcode.”
938—“Invalid Entry.”
939—“I’m sorry, the line is busy.”
940—“I’m sorry, the person you have called is not available.”
941—“Message sent.”
943—“Please enter your passcode.”

944—“Thank you for calling. If you know your party’s extension, you may enter it at any time. To access a voice mail box, press 6 plus the extension. To dial by name press 1. To reach the operator press 0. If you are calling from a rotary phone, please stay on the line and I’ll transfer you to the Operator.”

945—“Thank you for calling. If you know your party’s extension, you may enter it at any time. To access a voice mail box, press 6. To dial by name, press 1. To reach the Operator, press 0. If you are calling from a rotary phone, please stay on the line and I’ll transfer you to the Operator.”

946—This plays a generic on-hold music file and an on-hold courtesy message. It is assigned to the first Loop Tape.

Phrase List

Numbers	**Days/Months/Time**
Zero (says “zero”)	Monday
One (says “one”)	Tuesday
Two (says “two”)	Wednesday
Three (says “three”)	Thursday
Four	Friday
Five	Saturday
Six	Sunday
Seven	January
Eight	February
Nine	March
Ten	April
Eleven	May
Twelve	June
Thirteen	July
Fourteen	August
Fifteen	September
Twenty	October
Thirty	November
Forty	December
Fifty	AM
Sixty	PM
Seventy	
Eighty	
Ninety	
Hundred	
Teen	
Thousand	

System

A	Busy	Extension	Holding	Minus	Today
Account	Call	File	Invalid	Not	Transfer
Again	Completed	Files	Is	Number	Try
All	Delete	Forward	Later	Off	Up
And	Deleted	Forwarded	Leave	On	Voice
Answer	Disturb	Found	Lines	Please	Wait
Are	Do	Group	Mail	Port	Waiting
Available	Enter	Hang	Message	Retry	Zone
Box	Error	Hold	Messages	Thank You	

SCHEDULED DISK OPTIMIZATION

The following section describes the procedure for setting up a scheduled event to run the Caldera® DR-DOS® disk optimization tools on the Telecor Voice Server (TVS) hard disk.

Warning! The Caldera DR-DOS **`chkdsk`** and **`diskopt`** commands replace Microsoft® ScanDisk and Microsoft® Defragmenter used with VS1 Software 2.7.14a and earlier. Do not use ScanDisk or Defragmenter with a TVS running DR-DOS. ScanDisk or Defragmenter may corrupt Telecor Voice Servers running DR-DOS.

1. If needed, create a batch file named **`user.bat`** in the **`c:\ops`** directory. Add the following lines to the end of the file in the order shown.

   ```
   if exist c:\temp.dat chkdsk /f /b
   if exist c:\temp.dat diskopt /o /m1
   if exist c:\temp.dat del c:\temp.dat
   ```

2. Use the **`dskmaint.cmd`** in the **`c:\ops\cfg`** directory. It should contain the following commands:

   ```
   copy c:\ops\cfg\system.dat c:\temp.dat
   shutdown 0
   ```

3. Create a second command script named **`night.cmd`** in the **`c:\ops\cfg`** directory containing the following command:

   ```
   cload night.cfg
   ```

4. Create a scheduled event for the **`dskmaint.cmd`** at 1:00 a.m. Monday morning (or whenever you want the optimization to run).

5. Create a scheduled event for **`night.cmd`** to run 1 minute after **`dskmaint.cmd`** (for example 1:01 a.m.).

   - This is important because it enables the system to reload the night configuration, and then return to normal operation.

STATION MESSAGE DETAIL RECORDING (SMDR)

Station Message Detail Recording (SMDR) data provides a record of activities on the VS1 telephone system. Four different SMDR data files are generated by the Telecor Voice Server (TVS): Detail, Summary, ACD, and Day. The comma-delimited data from these reports can be imported directly into a spreadsheet or database, or used with a call accounting program to evaluate call performance.

- **Detail SMDR data** provides more detailed information for each call, breaking the call into segments and recording every change in action on a phone. It can be quite extensive, but is a good way to find out exactly what is happening with a phone. Only calls which access CO Lines are recorded.

- **Summary SMDR data** provides details about calls, and has one record for each call and each transferred call. This information is recorded when the call is ended or transferred. Only calls which access CO Lines are recorded.

- **ACD SMDR data** provides detailed information about all ACD activities, including Queue Warnings, Terminated Calls, Wrap-up and so on.

- **Day SMDR data** provides the names and numbers assigned to each port on the system. Information is recorded every time a name, extension or port type is changed.

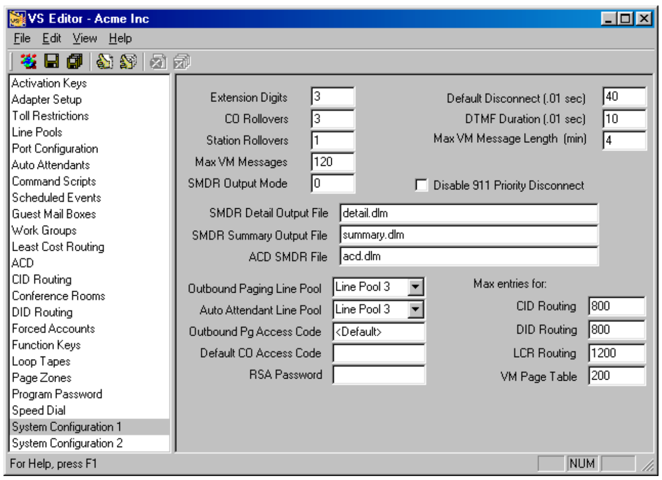
Designating SMDR Output Files

For the TVS software to record the SMDR information, output files must exist to which the information can be written. By default, output files exist for Summary SMDR (**summary.dlm**), ACD SMDR (**acd.dlm**), and Day SMDR (**day.dlm**) data. To add an output file for Detail SMDR data, or to rename the **summary.dlm** and **acd.dlm** output files, the TVS must be configured with the VS1 Editor application.[3] Complete the following steps:

1. In the VS1 Editor application, select **System Configuration 1**.

2. In the **System Configuration 1** pane, the following text boxes are displayed:

   SMDR Detail Output File
   SMDR Summary Output File
   ACD SMDR File



System Configuration 1 pane

3. In the **SMDR Detail Output File** text box, type the name of the output file. Telecor recommends **detail.dlm**. If required, rename the **summary.dlm** and **acd.dlm** files. Any valid path and name can be entered. For example: **c:\ops\detail.dlm**. (If no path is included the output files are stored in the **c:\ops** directory.) The name can be no longer than eight characters and must end with the **.dlm** extension.

Note	If the SMDR Output File text lines are left blank, no SMDR data is recorded.

Backing up SMDR Output Files

When the system is turned on, the output **.dlm** files begin to collect SMDR data. This can cause the output files to become quite large. By default, a Command Script has been created for each day of the week that backs up the SMDR data from the output files for that particular day. Data that are backed up include Detail, Summary and ACD SMDR. The Day SMDR data is not backed up. After backing up the SMDR data, the original **.dlm** files are deleted to begin a fresh recording. This keeps the ongoing original `.dlm` file down in size.

[3] The **day.dlm** output file is automatically created by the TVS and cannot be renamed. The **day.dlm** file is stored in the **c:\ops\cfg** directory.

Below is a list of the daily Command Scripts that have been created to backup SMDR data. A Command Script that backs up SMDR data for a particular day is run by a Scheduled Event at 12:01 AM on the next day. For example, the SMDRMON Command Script is run at 12:01 AM Tuesday to backup Monday's SMDR data.

SMDRMON – Backs up SMDR data to summary.mon, detail.mon, and acd.mon files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRTUE – Backs up SMDR data to summary.tue, detail.tue, and acd.tue files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRWED – Backs up SMDR data summary.wed, detail.wed, and acd.wed files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRTHU – Backs up SMDR data to summary.thu, detail.thu, and acd.thu files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRFRI – Backs up SMDR data to summary.fri, detail.fri, and acd.fri files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRSAT – Backs up SMDR data to summary.sat, detail.sat, and acd.sat files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

SMDRSUN – Backs up SMDR data to summary.sun, detail.sun, and acd.sun files, and then deletes the summary.dlm, detail.dlm and acd.dlm files.

Backup files are stored in the **c:\ops** directory.

Command Script Sample to Copy SMDR Files on Tuesday

Below are the commands that are included in the SMDRMON Command Script. Note that **summary** and **acd** data are kept for a maximum of four weeks.

copy summary.mo3 summary.mo4
copy acd.mo3 acd.mo4

copy summary.mo2 summary.mo3
copy acd.mo2 acd.mo3

copy summary.mon summary.mo2
copy acd.mon acd.mo2

copy summary.dlm summary.mon
copy acd.dlm acd.mon

copy detail.dlm detail.mon

del detail.dlm
del summary.dlm
del acd.dlm

Gaining Access to SMDR Backup Files

To use the information provided by the SMDR files, it is necessary to copy the backup files from the TVS. After the files are copied, it can then be imported into another application and formatted for use. Three different methods can be used to copy the backup files.

- Copying the files onto a disk
- Copying files through a network connection
- Copying files through the Tel-Site system management application.

Copying SMDR Backup Files to a Disk at the TVS

1. Insert a disk into Drive A of the TVS.

2. At the TVS Command prompt, use the following command to copy a backup file:

   copy {source} {destination} - with [source] representing the backup file and [destination] representing the path and copied file. Destination names must be no longer than eight characters. It is recommended that destination names use the **.csv** extension as most spreadsheet programs will automatically recognize it as a comma-delimited file.

   For example:

```
copy summary.mon a:\summary.csv
copy summary.mo2 a:\summ02.csv
copy acd.mon a:\acd.csv
copy acd.mo2 a:\acdmo2.csv
copy detail.mon a:\detail.csv
```

Note: If the [source] file is not in the default **c:\ops** directory, the path then must be specified.

Note: Because the **day.dlm** is not backed up, and is stored in a different directory, the following command is used: `copy c:\ops\cfg\day.dlm a:\day.csv`

3. Press ENTER on the keyboard to copy the files onto disk.

Note If you receive a File Not Found message when you press ENTER, type `dir` at the TVS Command prompt for a listing of backup files on the TVS.

Copying SMDR Files through a Network

To copy SMDR files through a network, the TVS must have a network card installed and be attached to a network. In addition. for this option to work, the **System Configuration 2** pane must be properly configured in the VS1 Editor application. *See "Configuring the 10Base-T Network Interface Card" in the Hardware section.*

1. At the TVS Command prompt, use the following command to copy a backup file:

   copy {source} {destination} - with [source] representing the backup file and [destination] representing the path and copied file. Destination names must be no longer than eight

characters. It is recommended that destination names use the **.csv** extension as most spreadsheet programs will automatically recognize it as a comma-delimited file.

For example:

```
copy summary.mon net:summary.csv
copy summary.mo2 net:summ02.csv
copy acd.mon net:acd.csv
copy acd.mo2 net:acdmo2.csv
copy detail.mon net:detail.csv
```

Note: If the [source] file is not in the default **c:\ops** directory, the path then must be specified.

Note: Because the **day.dlm** is not backed up, and is stored in a different directory, the following command is used: `copy c:\ops\cfg\day.dlm net:day.csv`

Copying SMDR Files through Tel-Site

1. Click the **Explorer window** button on the toolbar of the **Connection** window.

2. In the **All Folders** window, double-click the **Ops** folder.

3. Select the backup file to copy, and then select the **Download** option from the **Explorer** menu.

   The file is copied to the **Ops** folder of the Customer Site folder on the local drive. You can copy the file using the Windows® Explorer, or open it directly within an application. The name of the copied file can be changed, as long as the name does not exceed eight characters. It is also recommended that the file use the **.csv** extension as most spreadsheet programs will automatically recognize it as a comma-delimited file.

Interpreting Summary SMDR Data

By default, the Summary SMDR data is recorded in 15 comma-delimited fields. If Software Protection is enabled, a 16th row is recorded. Each row represents one call and each field records a specific aspect of a call.

Field	Description	Type	Maximum Width
1	Call Sequence Number	Numeric	10
2	Call Type (0=out, 1=in, 2=disa)	Numeric	1
3	Call Reason	Numeric	2
4	CO Port Used	Numeric	4
5	Extension Port Used	Numeric	4
6	Date of Call	Date	10
7	Time of Call	24-Hour Format	8
8	Total time of Call (decimal minutes)	Numeric	10
9	Time On Hold (decimal minutes)	Numeric	10
10	Touch Tones Dialed	Character	40
11	Message Sent to Phone	Character	40
12	Account Code	Character	16
13	Caller ID Number	Character	40
14	Caller ID Name	Character	33
15	Called Number	Character	17
16	Software Protection (if enabled)	Numeric	10

Summary SMDR data imported into a spreadsheet

Call Sequence	Call Type	Call Reason	CO Port	Ext. Port	Date	Time of Call	Total Time	Time on Hold	Touchtones Dialed	Message sent to Phone	Account Code	Caller ID Number	Caller ID Name	Called Number	Software Protection
166688	1	3	3	7	5/21/2003	14:10:44	0.563	.228				5640801	Susan Richmond		
166688	1	1	3	12	5/21/2003	14:11:16	0.256	0				5640801	Susan Richmond		
166701	0	1	3	14	5/21/2003	14:16:06	0.049	0	18002E+10	18002E+10					
166705	0	1	3	14	5/21/2003	14:16:22	0.25	0	5556676	5556676					

Note Field headers are not included in SMDR data, and must be added manually, as shown in the table above.

For Summary SMDR, the Call Sequence Number (Field 1) is a number assigned to each call and each transferred call. This information is recorded when the call is ended or transferred. The Call Sequence Number can be repeated in multiple rows, representing the initial call and each subsequent transfer.

The Software Protection field relates to the **SMDR Output Mode 0/1** setting of the **System Configuration 1** pane of the VS1 Editor. By default, the setting is 0. If set to 1, Software Protection data is recorded. This will record the last five digits of the Host Adapter Card for the system, which can be used for call accounting software.

Codes for Call Reason (Field 3) of Summary SMDR

00 = CO Line dropped current
01 = Disconnect received
02 = Call placed on hold
03 = Transfer
04 = CO Connect (answer)
05 = Reconnect from hold
06 = CO Acquired and connected
07 = CO Line initial ring
08 = Ring no answer
09 = CO error (line reset)
10 = Current drop while waiting for extension
11 = CO cleared by system console
12 = CO cleared while ringing
13 = DISA account code entered
14 = DISA dial string
15 = Invalid DISA account - disconnect
16 = Invalid DISA function
17 = Abort/Stop current call - DISA
18 = Flash

Additional Port Numbers Used in Extension Port (Field 5)

1000 = Voice Mail for Extension, i.e. 1110 for extension 110
3000 = Auto Attendant
900 = ACD

Interpreting Detail SMDR Data

The Detail SMDR data is recorded in 15 comma-delimited fields. If Software Protection is enabled, a 16th row is recorded. Each field records a specific aspect of a call, and is limited to type of data and width of data recorded.

Field	Description	Type	Maximum Width
1	Call Sequence Number	Numeric	10
2	Call Type (0=out, 1=in, 2=disa)	Numeric	1
3	Call Reason	Numeric	2
4	CO Port Used	Numeric	4
5	Extension Port Used	Numeric	4
6	Date of Call	Date	10
7	Time of Call	24-Hour Format	8
8	Length of Call (days)	Numeric	5
9	Length of Call (decimal minutes)	Numeric	10
10	Touch Tones Dialed	Character	40
11	Message Sent to Phone	Character	40
12	Caller ID Number	Character	40
13	Caller ID Name	Character	33
14	Account Code	Character	16
15	Called Number	Character	17
16	Software Protection (if enabled)	Numeric	10

Detail SMDR data imported into a spreadsheet

Call Sequence	Call Type	Call Reason	CO Port	Ext. Port	Date	Time	Length-Days	Length-Minutes	Touchtones Dialed	Message sent to Phone	Caller ID Number	Caller ID Name	Account Code	Called Number	Software Protection
166688	0	1	3	0	5/21/2003	14:13:26	0	0	8887936	NC					
166688	0	0	3	0	5/21/2003	14:13:48	0	363	8887936	NC					
166692	0	1	3	0	5/21/2003	14:13:59	0	0	5551234	NC					
166692	0	0	3	0	5/21/2003	14:14:14	0	256	5551234234	NC					
166701	0	1	3	0	5/21/2003	14:16:06	0	0	18002E+10	NC					
166701	0	1	3	0	5/21/2003	14:16:08	0	49	18002E+10	NC					

Note Field headers are not included in SMDR data, and must be added manually, as shown in the table above.

For Detail SMDR, the Call Sequence Number (Field 1) is a number assigned to each call. The Call Sequence Number can be repeated in multiple rows, representing different aspects of a single call. When tracking a call in a report, note all occurrences of the Call Sequence Number.

The Software Protection field relates to the **SMDR Output Mode 0/1** setting of the **System Configuration 1** pane of the VS1 Editor. By default, the setting is 0. If set to 1, Software Protection data is recorded. This will record the last five digits of the Host Adapter Card for the system, which can be used for call accounting software.

Codes for Call Reason (Field 3) of Detail SMDR

00 = CO Line dropped current
01 = Disconnect received
02 = Call placed on hold
03 = Transfer
04 = CO Connect (answer)
05 = Reconnect from hold
06 = CO Acquired and connected
07 = CO Line initial ring
08 = Ring no answer
09 = CO error (line reset)
10 = Current drop while waiting for extension
11 = CO cleared by system console
12 = CO cleared while ringing
13 = DISA account code entered
14 = DISA dial string
15 = Invalid DISA account - disconnect
16 = Invalid DISA function
17 = Abort/Stop current call - DISA
18 = Flash

Additional Port Numbers Used in Extension Port (Field 5)

1000 = Voice Mail for Extension, i.e. 1110 for extension 110
3000 = Auto Attendant
900 = ACD

Interpreting ACD SMDR Data

The ACD SMDR data is recorded in 13 comma-delimited fields. If Software Protection is enabled, a 14th row is recorded. Each field records a specific aspect of a call, and is limited to type of data and width of data recorded.

Field	Description	Type	Maximum Width
1	ACD number	Numeric	6
2	Date of Call	Date	10
3	Time of Call	24-Hour Format	8
4	Agent Port Number	Numeric	6
5	Agent Account Code	Numeric	6
6	Agent Priority	Numeric	6
7	Caller Port Number	Numeric	6
8	Call Sequence Number	Numeric	10
9	Caller ID Number	Character	15
10	Action Code	Numeric	2
11	Related Call Sequence	Numeric	10
12	Called Number	Character	18
13	Message Sent	Character	40
14	Software Protection (if enabled)	Numeric	10

Note Telecor Voice Server Software records 10 characters (mm/dd/yyyy) in Date of Call (Field 2) in the ACD SMDR data. If your call accounting software requires only eight (8) characters (mm/dd/yy) you can edit the `pbx` line of the `startpbx.bat` file to read `pbx -27`.

ACD SMDR data imported into a spreadsheet

ACD #	Date of Call	Time of Call	Age nt Port	Agent Account Code	Agent Priority	Caller Port	Call Sequence	Caller ID #	Action Code	Related Call Sequence	Called #	Message Sent	Software Protection
1	05/21/2003	14:14:58	0	0	0	3000	166695		8	0	421	Call from AA1	
1	05/21/2003	14:14:58	0	0	0	3000	166695		26	0	421	Call from AA1	
1	05/21/2003	14:14:58	0	0	0	1	166693	107	24	1666695	401	Call from AA1	
1	05/21/2003	14:15:13	1	3	0		2	14	25	0			
1	05/21/2003	14:15:20	12	1001	1	6	166693	107	22	0	401	Call from AA1	
1	05/21/2003	14:15:22	12	1001	1	13	166693	107	23	0	401	Call from AA1	
1	05/21/2003	14:15:22	12	1001	1	13	166693	107	11	0	401	Call from AA1	
1	05/21/2003	14:16:48	12	1001	1	13	166693	107	4	0	401	Call from AA1	
1	05/21/2003	14:16:56	0	0	0	13	166706	101	8	0	421	IN:Walter O'Rile	
1	05/21/2003	14:16:59	1	3	0	7	2	3	25	0			
1	05/21/2003	14:17:14	0	0	0	7	166706	101	7	0	421	IN:Walter O'Rile	
1	05/21/2003	14:17:14	300	0	0	7	166706	101	18	0	421	$AA1.BIN	
1	05/21/2003	14:17:15	0	3	0	7	2	0	25	0			
1	05/21/2003	14:17:25	0	0	0	3000	166709		8	0	421	Call from AA1	
1	05/21/2003	14:17:25	0	0	0	3000	166709		26	0	421	Call from AA1	

Note Field headers are not included in SMDR data, and must be added manually, as shown in the table above.

The Call Sequence Number (Field 8) is a number assigned to each call. In the ACD SMDR, the Call Sequence Number can be repeated in multiple rows, representing different aspects of a single call. When tracking a call in a report, note all occurrences of the Call Sequence Number.

The Software Protection field relates to the **SMDR Output Mode 0/1** setting of the **System Configuration 1** pane of the VS1 Editor. By default, the setting is 0. If set to 1, Software Protection data is recorded. This will record the last five digits of the Host Adapter Card for the system, which can be used for call accounting software.

Action Codes for Field 10 of ACD SMDR Data

01 = Logon to ACD
02 = Logoff, Manual
03 = Logoff, Automatic (no answered timeout)
04 = Call Terminated while connected to Agent
07 = Caller Pressed 0, Transfer to AA
08 = New Call
11 = Call connected to Agent
13 = Hang up, called disconnected while in Queue
14 = Warning Threshold 1 (Too many calls)
15 = Warning Threshold 2 (Over time)
16 = Warning Threshold 3 (No Agents)
18 = Call Transferred (New Agent has call)
19 = Wrap-up Start
20 = Wrap-up End
21 = ACD Full (Over 30 calls in queue)
22 = Call Agent - 1st Ring
23 = Transfer Complete
24 = New Call Transferred
25 = Queue Status (see table)
26 = Consultation Call Terminated

Interpreting ACD Queue Status SMDR Data

When the Action Code (Field 10) of the ACD SMDR is 25, the data in that row is a report of the ACD Queue Status. ACD Queue Status is sent to the `acd.dlm` at regular intervals (in seconds), as set on the **Queue Status Time** text line of the **ACD Configuration** pane of the VS1 Editor. The ACD Queue Status SMDR data is recorded in 11 comma-delimited fields. If Software Protection is enabled, a 12th row is recorded.

Field	Description	Type	Maximum Width
1	ACD number	Numeric	2
2	Queue Status Date	Date	10
3	Queue Status Time	24-Hour Format	8
4	Number of Calls in Queue	Numeric	4
5	Number of Stations Logged On	Numeric	4
6	Number of Calls in Service	Numeric	4
7	Total Calls	Numeric	10
8	Total Hang ups	Numeric	10
9	Age of Oldest Call	Numeric	10
10	Action Code	Numeric	2
11	Related Call Sequence	Numeric	10
12	Software Protection (if enabled)	Numeric	6

ACD Queue Status SMDR data imported into a spreadsheet

ACD #	Queue Status Date	Queue Status Time	Number Calls in Queue	Number of Stations Logged On	Number of Calls in Service	Total Calls	Total Hangups	Age of Oldest Call	Action Code	Related Call #	Software Protection
1	05/21/2003	14:15:00	1	3	0	6	2	14	25	0	
1	05/21/2003	14:15:20	1	3	0	7	2	3	25	0	
1	05/21/2003	14:15:40	0	3	0	7	2	0	25	0	
1	05/21/2003	14:16:00	0	3	0	8	2	0	25	0	
1	05/21/2003	14:16:20	0	3	0	9	3	0	25	0	

Note Field headers are not included in SMDR data, and must be added manually, as shown in in the table above.

Interpreting Day SMDR Data

The Day SMDR data is recorded in five comma-delimited fields into the **day.dlm** file. Each field records the following information about a port: extension description, device type, extension number and call pick-up group. This information is used by a Call Accounting program to generate SMDR reports (Detail, Summary and ACD) with sensical information rather than just the port number.

Field	Description	Type	Maximum Width
1	Port Number	Numeric	4
2	Extension Description	Character	30
3	Device Type	Numeric	2
4	Extension Number	Numeric	4
5	Group	Numeric	4

Day SMDR data imported into a spreadsheet

Port Number	Extension Description	Port Type	Extension Number	Call Pick-upGroup
1	CO Line 1	2	-1	
2	CO Line 2	2	-1	
3	CO Line 3	2	-1	
4	CO Line 4	2	-1	
5	Receptionist	2	0	
6	Norm Stenger	2	100	
7	Walter O'Riley	1	101	1
8	Board Room	6	102	1

Codes for Port Type (Field 3) of Day SMDR Data

0=No Connection
1=Standard Phone
2=CO Line
3=3 Line Feature Phone
4=Reserved 1
5=Call
6=Modem, Fax
7=Display Phone
8=Attendant
9=Connect/CTIM
10=Connect/PCOM

Note **Call** (Code 5) and **3 Line Feature Phone** (Code 3) are reserved for backwards compatibility.

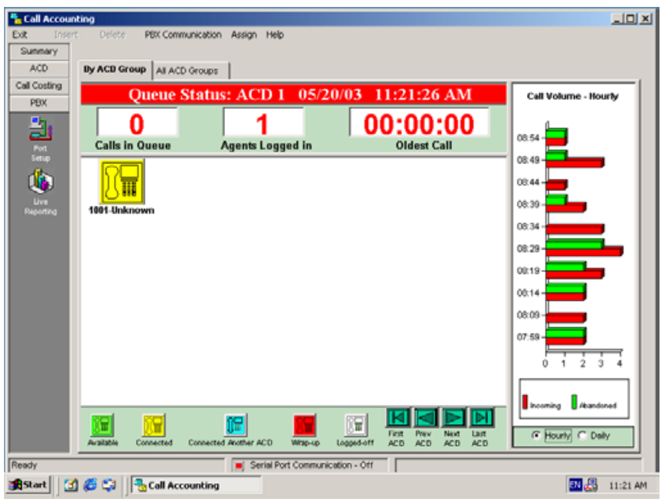
VS CALL ACCOUNTING

The VS Call Accounting application offers an easy-to-use Window® interface for viewing SMDR data from the VS1 System. By connecting a PC to the TVS via a serial cable, the user has the ability to view imported files or conduct real-time monitoring of both ACD and Summary SMDR data.

Reports can be generated in a variety of tabular and graphical formats, including bar graphs, line graphs and pie graphs. Tabular reports are easily customizable in terms of what fields are selected for display. For billing purposes, a Call Costing add-on module enables fixed and incremental rates (in seconds and minutes) to be set for each area code.

Requirements

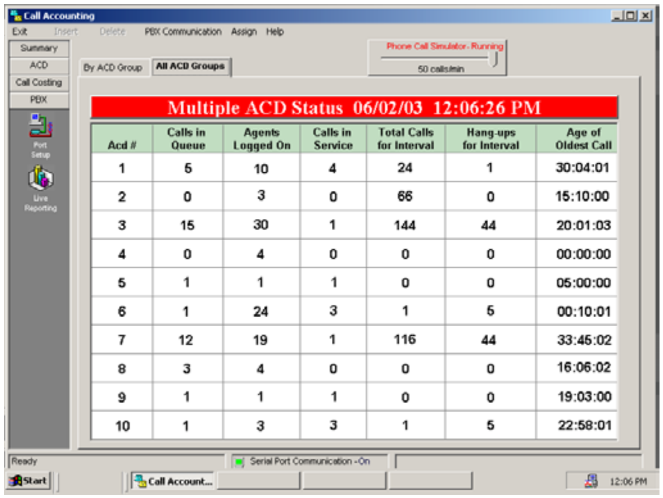
- Pentium 120 PC or faster, 32 megabyte (MB) of RAM, 1 gigabyte (GB) hard drive
- Microsoft® Windows® 2000 and Windows® XP operating systems
- Open COM port with dedicated IRQ if performing live monitoring
- 15-inch color monitor recommended; 640x480 mode



ACD SMDR Live Reporting

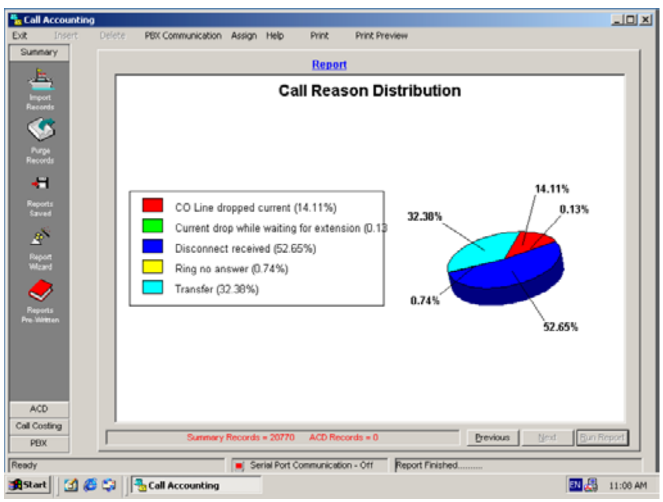
The ACD Live monitoring screen displays real-time activity of a selected ACD, including number of available agents (logged on), the number of agents currently on calls, the number of agents in "wrap-up," number of calls in queue, and the age of oldest call in queue.

Multiple ACD Status

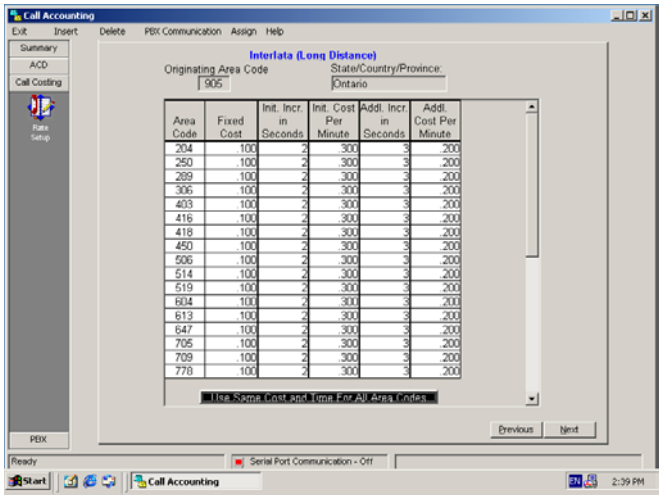


The Multiple ACD window provides an up-to-date overview of the status of each ACD.

Pre-written Summary SMDR data



SMDR data can be viewed in a variety of tabular and graphical formats, including bar graphs, line graphs and pie graphs.



Call Costing

The Call Costing screen allows fixed and incremental rates (in seconds and minutes) to be set for each area code, allowing individual billing to be charged to each extension.

VERIFYING CO DISCONNECT SIGNALS

The VS1 telephone system requires Loop Start CO lines with cutoff on disconnect (COD) or disconnect supervision for proper operation. A Disconnect Signal is simply a 500 to 900 millisecond (ms) drop in loop current provided by the telephone company to signal the end of the current call. The VS1 phone system relies on these Disconnect Signals to properly terminate CO calls that are connected to voice channels (i.e. Auto Attendants, Voice Mail, and ACDs).

If you suspect the local telephone company is not providing disconnect signaling, follow these steps to verify the CO Disconnect Signal:

Note This test should be performed while system activity is at a minimum.

1. In the **System Configuration 1** window, type **999** in the **Default Disconnect (.01 sec)** text box. This change is required to properly perform the disconnect test.

2. Verify that the **Disconnect** parameter is set to –1 in every analog (non-T1) CO **Port Configuration** edit window.

3. Restart the phone system.

4. At the Telecor Voice Server (TVS) Command prompt type **`set debug=7`** and press ENTER. Next, type **`set errorlog=error.log`** and press ENTER.

Note If you are performing this test on a Telecor Voice Server Model 200 (PV-CSU-200), connect to the system through the Tel-Site system management application and issue the commands from the **Terminal** window.

5. Have an assistant place an inbound call on one of the CO lines. Answer the call from a station set. When you are connected to the outside caller, instruct them to hang up.

   - Important: Do not hang up your station set for 15 seconds.

6. At the TVS Command prompt type **`clear [port #]`** using the port number that the call rang in on, and press ENTER. This releases the call if the phone system hasn't disconnected already.

7. Repeat steps 5 and 6 at least four times.

8. In the **System Config 1** pane set the **Default Disconnect** back to **40**, and then reset the system to put this change into effect.

9. At the TVS Command prompt type **`copy error.log a:error.log`** to copy the file **`error.log`** from the **`c:\ops`** directory to a disk in the a: drive. If you are connected to the system through Tel-Site, use the **Explorer** window to download **`error.log`** to your local system.

10. Open the **`error.log`** file in a text editor such as DOS Edit or Notepad. Search for lines of text that contain the words "CO:Flash" (i.e. CO:Flash 820 mS[1]) and make note of the values immediately after the word "Flash".

    - These values are the length of the disconnect signals that the phone system recorded on the CO port.

To calculate the proper Disconnect Signal setting for the system, first discard any values that are less than 450 mS. Select the smallest remaining value, subtract 20 mS from that value, and then divide the difference by 10. Use this value as the new Disconnect Signal.

For example, if the following Disconnect Signal values were recorded:

CO :Flash 800 mS[1]
CO :Flash 820 mS[1]
CO :Flash 750 mS[1]
CO :Flash 325 mS[1]
CO :Flash 950 mS[1]

Discard values less than 450 mS. Subtract 20 mS from the smallest remaining value, which is 750 mS. Divide the resulting difference, 730 mS, by 10. The correct Disconnect Signal setting for this system is 73.

Note	If none of the recorded values are greater than 400 mS, disconnect supervision is not being provided. Contact the local telephone company and request that COD service be added to the lines.

T1 INTERFACE CARD CONFIGURATION

After the T1 Interface Card has been installed in the TVS, it then must be configured using the T1 Edit program in the Tel-Site system management application. After this has been completed, the VS1 Editor program must be used to enable the VS1 phone system to recognize the T1 Interface Card.

T1 Interface Card Parameter Setup Using Tel-Site

1. Use Tel-Site to gain access to the Telecor VS1 phone system.

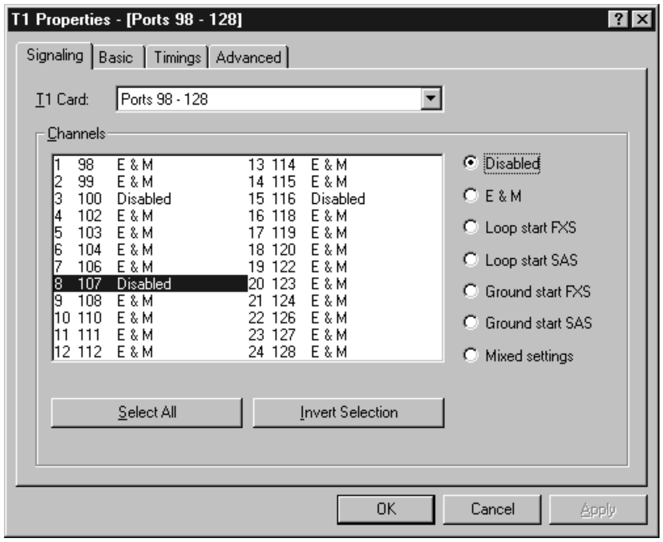
2. Click the **T1 Edit** button in the **Configuration** window of Tel-Site to open the **T1 Properties** dialog box. The **T1 Properties** dialog box contains four tabs: **Signaling**, **Basic**, **Timing** and **Advanced**.

   - Use the **Signaling** tab to set the signaling type for each channel on the T1 Interface Card based on the information provided by the Service Provider.
   - Select the **Basic** tab to set the framing for the T1 Interface Card. Framing is used by the T1 Interface Card to synchronize with the Service Provider. If the correct framing is not selected, the T1 Interface Card cannot carry voice information. The framing required and its associated configuration is determined by the Service Provider.
   - The **Timings** tab is used to set the timing parameters for the T1 Interface Card.
   - The settings on the **Advanced** tab rarely require adjustment and should only be changed at the direction of Telecor Technical Support.

3. Select the **Signaling** tab on the **T1 Properties** dialog box. Use the **T1 Card** text box to select the T1 Interface Card for which you want to set signaling types. All ports and their signaling status for the T1 Interface Card selected are listed in the **Channels** group box. You can set each port individually; click the **Select All** button to set all ports; or select ports you do not want to set, and then click the **Invert Selection** button to select all other ports. To change the signaling status for a port, click one of the following option buttons in the **Channels** group box:

   - **Disabled**—Select Disabled when this port of the T1 Interface Card is not used.
   - **E&M**—E&M is the preferred method of signaling for DID ports. E&M signaling must be set when DID, DNIS or ANI service is provided by the Service Provider.
   - **Loop Start FXS** or **Loop Start SAS**—These signaling methods are not recommended because a disconnect signal is not generated by the Service Provider. The use of FXS versus SAS depends on the Service Provider. FXS stands for Foreign Exchange Station. SAS stands for Special Access Station.
   - **Ground Start FXS** or **Ground Start SAS**—Use these signaling methods when DID or DNIS service is not being used. The use of FXS versus SAS depends on the Service Provider. FXS stands for Foreign Exchange Station. SAS stands for Special Access Station.
   - **Mixed settings**—Mixed settings Enables you to select mixed settings.

Note Reserved ports are not displayed and their settings cannot be changed.

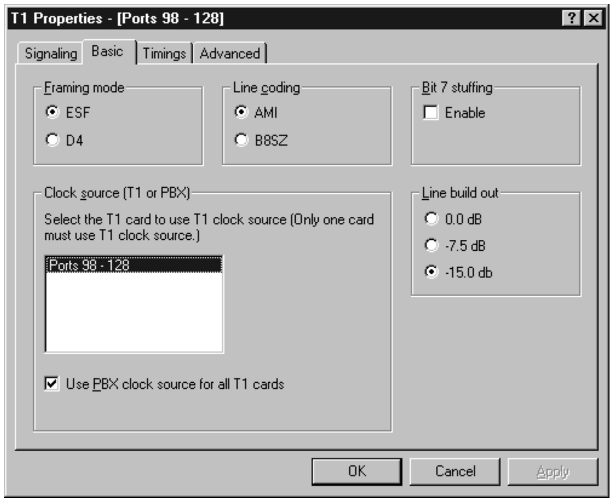


Signaling tab

4. Select the **Basic** tab on the **T1 Properties** dialog box to set up the framing for the T1 Interface Card. You have the following options:

   - **Framing mode**—ESF is the preferred framing mode; however, many Service Providers only support the D4 framing mode.
   - **Line coding**—B8ZS is the preferred Line Coding when using ESF framing. AMI is the preferred Line Coding when using D4 framing.
   - **Bit 7 stuffing**—Some Service Providers may require you to enable Bit 7 stuffing when B8ZS is not used.
   - **Line build out—**This determines the strength of the transmitted signal from the T1 Interface Card to the Service Provider. There are three levels: 0.0 dB, –7.5 dB, and –15.0 dB. Set the Line build out to –15.0 dB, and increase to –7.5 dB or 0.0 dB if the Service Provider does not detect a transmitted signal from the T1 Interface Card.
   - **Clock source (T1 or PBX)**—It is important to set up the Clock source correctly. If only one T1 Interface Card is installed, and the system is connected to a Service Provider, select a Clock source of T1. If two T1 Interface Cards are installed, then select one card as the master, and set it as the T1 Clock source. The other T1 Interface Card must have the Clock source set to PBX. If more than one card has its Clock source set to T1, the phone system will not work.

Note	If both T1 Interface Cards have their Clock source set to PBX, the system will work but the T1 connection will be degraded. The default Clock source setting is PBX.



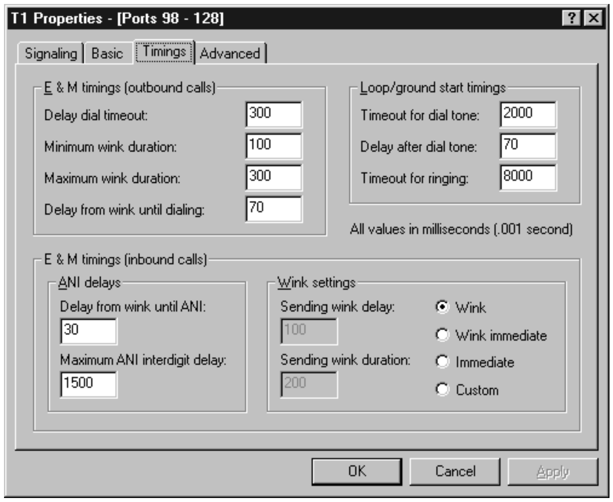
Basic tab

5. Select the **Timings** tab on the **T1 Properties** dialog box to set the timing parameters for the T1 Interface Card. The **Timings** tab is divided into four group boxes:

    - **E&M Timing (outbound calls)**—The **Delay dial timeout** sets the timeout for the PBX to detect the start of a wink signal; the **Minimum** and **Maximum wink duration** sets the minimum and maximum length for a wink signal from the T1 Service Provider; **Delay from wink until dialing** sets the delay after the T1 Service Provider wink signal until the PBX starts dialing.
    - **Loop/ground start timings**—The **Timeout for dial tone** sets the timeout for the PBX that is detecting dial tone from the T1 Service Provider; **Delay after dial tone** sets the delay that occurs when the T1 Service Provider asserts a dial tone until the PBX starts dialing; **Timeout for ringing** sets the timeout for the PBX detecting a loss of ring signal.
    - **ANI delays**—**Delay from wink until ANI** is used to set the delay that occurs after the PBX wink signal until the T1 Service Provider sends ANI/DID tones; **Maximum interdigit delay for ANI** sets the timeout for delay between ANI/DID tones from the T1 Service Provider.

Note The settings for E&M Timing (outbound calls), Loop/ground start timings, and E&M Timings (inbound calls) usually do not require any adjustments. Changes to the settings are necessary only when trouble occurs with the default settings.

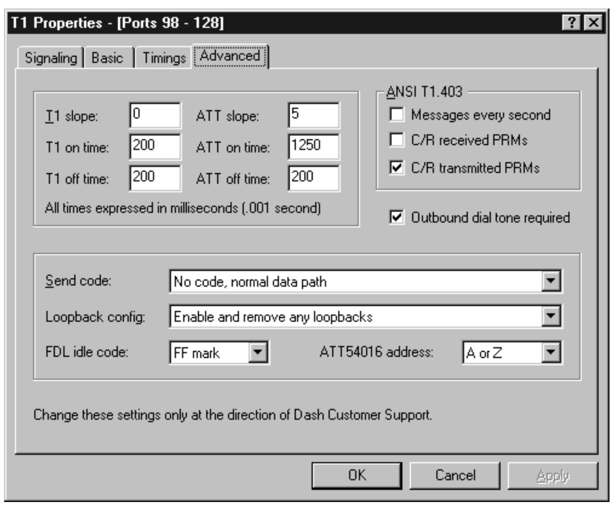
    - **Wink settings**—Select the appropriate option button based upon information provided by the T1 Service Provider, or use the **Custom** option button to set the wink timings manually.



Timings tab

6. Select the **Advanced** tab to make changes to any advanced settings.

Warning! Advanced settings rarely require adjustment and should only be changed at the direction of Telecor Technical Support.



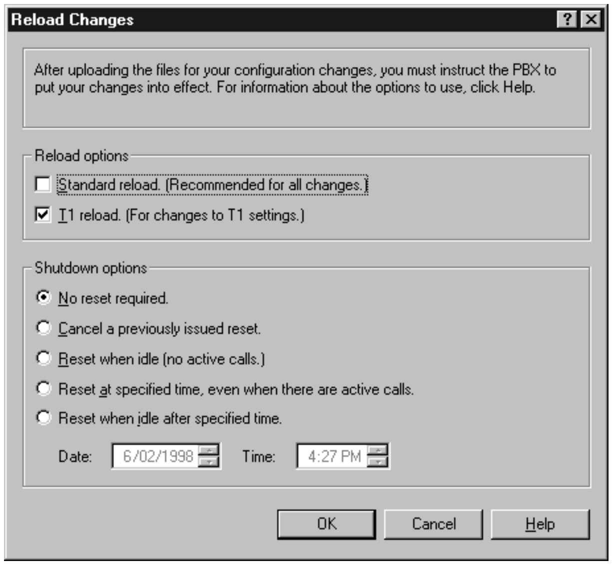
Advanced tab

7. Click **OK** to return to the **Configuration** window in Tel-Site and then click the **Reload Changes** button.

   - The Reload changes dialog box appears.

8. Select **T1 reload** in the **Reload options** group box, and then click **OK**.

Note It is not necessary to reset the system for changes made to T1 settings.



Reload Changes dialog box

T1 Interface Card Setup Using VS1 Editor

After using the T1 Edit program, the VS1 Editor configuration program must then be used to enable the VS1 phone system to recognize the T1 Interface Card.

Adapter Setup

Note If you already have a Host Adapter Card installed and a PEU is configured for Ports 96–128 or Ports 65–96, change the PEU port assignments before configuring a T1 Interface Card. PEUs and T1 Interface Cards cannot share the same port assignments. *For more information on changing PEU port assignments, see "Adapter Setup" in the VS1 Editor section.*

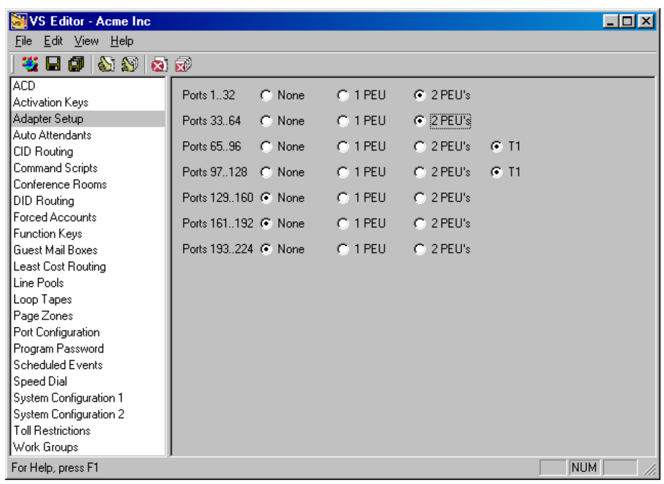
Complete the following steps:

1. Using the VS1 Editor configuration program, select the **Adapter Setup** pane.

2. For Ports 1..32 select if one or two PEUs are used. Repeat this step for additional ports.

3. If one T1 Card is installed, select the **T1** radio button in the **Ports 97..128** row. If a second T1 Card is installed, select the **T1** radio button in the **Ports 65..96** row.

4. Click the **Save** button in the toolbar.



Adapter setup pane

Configuring a T1 Port as a CO Line

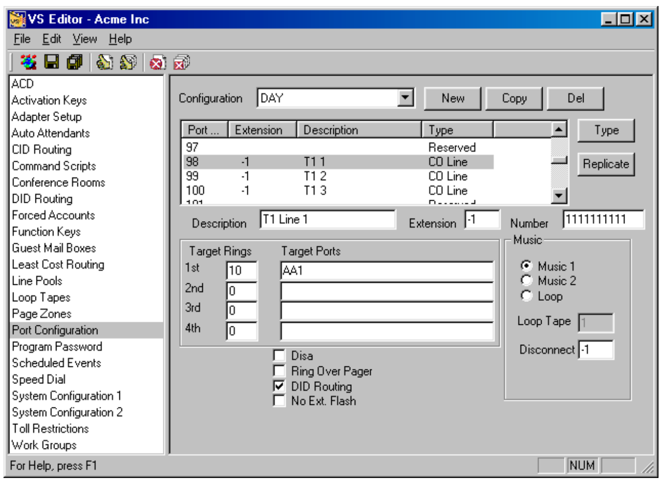
To configure a T1 port as a CO line, complete the following steps.

1. Using the VS1 Editor configuration program, click the **Port Configurations** pane.

2. In the **Port Configurations** pane, select the configuration you want to change. (For example, the DAY configuration).

3. Select the CO port you want to configure.

4. Beginning at Port 97 (for one T1 Interface Card) or Port 65 (for two T1 Interface Cards), every fourth port is reserved, and the others are unassigned. The reserved ports are a function of the T1 interface. You must configure each active channel on the T1 Interface Card as a CO line.



TI port configured as CO line

5. Select the unassigned port that you want to assign as a CO port, and then click **Type**. The **Port Type** window appears.

6. Select **CO Line**, and then click **OK** to return to the **Port Configuration** pane.

7. Repeat steps 5 and 6 for each port you want to set up as a CO Line.

8. To set up other configurations, repeat steps 2 through 7.

9. After all ports are assigned as CO Lines, click the **Save** button in the toolbar to save your changes.

Configuring a T1 Port for DID or DNIS

1. With a configuration selected in the **Port Configurations** pane, select a T1 port that you have assigned as a CO Line.

Note	If you have not assigned a port type to the port you selected, a warning message appears telling you to first assign a port type.

2. Enter the following information in the corresponding text boxes:

- **Description**: Enter a description for the T1 line.
- **Extension**: Enter an extension. (Enter **–1** if no extension is assigned)

- **Number**: Enter the digit **1** ten times, which function as placeholders.

Note A CO port configured for T1 service cannot be set up for both DNIS and DID.

3. Set up a default first ring target. This is important because if a call cannot be routed using the DID Routing table, it is routed based on the target rings and target port.

   - Under **Rings**, set the CO port to ring the first target for 10 rings.
   - Under **Target Ports**, set the CO port to ring the first Auto Attendant (AA1).

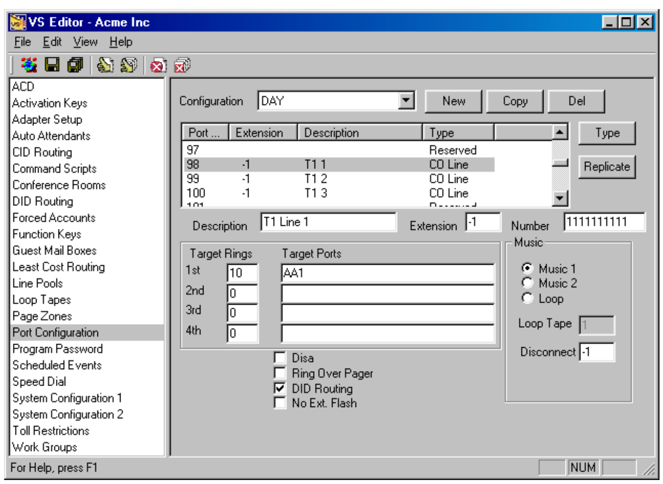
4. Check the **DID Routing** box.

5. To set up additional ports with DID or DNIS routing, repeat the above steps *or* use the **Replicate** button and follow the on-screen instructions.

6. Set up additional configurations by following steps 1 through 5.

7. After all CO ports are configured, click the **Save** button in the toolbar to save your changes.

Note When using ANI (Automatic Number Identification) only on the T1 port, **DID Routing** must remain unchecked.



TI Port configured for DID or DNIS

Setting Up a DID or DNIS Routing Table

To assign a block of numbers for DID routing, complete the following steps.

1. Select **DID Routing** in the Tree Control display.

   - The **DID Routing** pane appears.

2. Click **Add**.

3. In the **DID Entry** group box, enter the following information in the corresponding text boxes.

   Description: Enter a description for the block of numbers. The description is a 16-character message that is attached to the call. This message appears on all station equipment with a display.

Start Num: This box must contain 7 digits. Enter the DID start number of your block of numbers preceded by the necessary amount of placeholders. Telecor recommends using 1 for the placeholder since it is then obvious that they are not part of the routing number.

End Num: This box must contain 7 digits. Enter the DID end number of your block of numbers preceded by the necessary amount of placeholders. Telecor recommends using 1 for the placeholder since it is then obvious that they are not part of the routing number.

Example (for a 4-digit DID): Start Number 1114001 **End Number** 1114100

Start Time: Enter the start time that you want this block of numbers to recognized. Use standard military time format.

End Time: Enter the end time that you want this block of numbers to recognized. Use standard military time format.

Note: The **End Time** must not come before the **Start Time**.

Days: Select the days of the week that you want this block of numbers to be recognized.

4. Specify the **Target Ext** and **Target Mode**.

Target ext: Enter the target extension for the DID Routing Entry.

Fixed Target Mode: means that *all* numbers in the block are targeted to the same extension. For example, assigning 1114001–1114100 to Extension 103 means that 4001 goes to Extension 103; 4002 goes to Extension 103, and so on.

Note: DNIS numbers are usually assigned as individual numbers and not in related blocks. DNIS numbers can only be routed in the Fixed Target Mode.

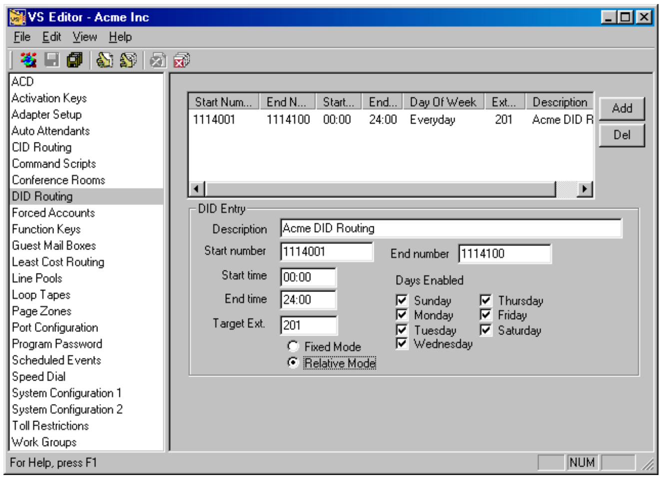
Relative Target Mode: means that *each* number in the block is targeted to a specific, individual extension. For example, if the Start Num is 1114001 and the Target Extension is 101, the number 4001 routes to extension 101, 4002 routes to extension 102, 4010 routes to Extension 110, and so on.

- To assign one number to a specific extension, type the same number in the **Start Number** and **End Number** text boxes. In the **Target Ext** text box enter the target extension, and then select **Fixed** as the **Target Mode**.

5. Click the **Save** button in the toolbar.

- The settings for the DID/DNIS Routing Table appear in the list box above.

6. Validate the table by clicking the **Validate this Pane** button in the toolbar.



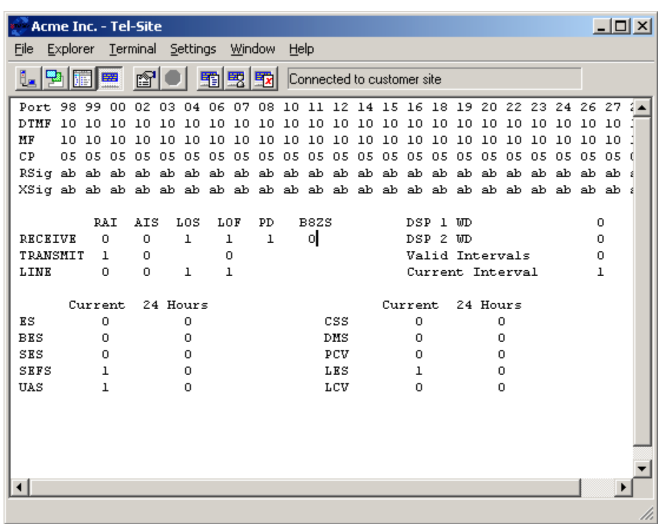
DID Routing pane

T1 Interface Card Diagnostics

The **`t1 stat`** command can be issued from the TVS Command prompt in **Terminal** window of Tel-Site or from the TVS Command prompt. This command provides a status screen displaying diagnostic information for the specified T1 Interface Card. Refer to the screen below.

- Issue the **`t1 stat 0`** command to display information on the first T1 Interface Card.
- Issue the **`t1 stat 1`** command to display information on the second T1 Interface Card.

Warning! Do not issue the **`t1 stat`** command on both the TVS and in the Terminal window of Tel-Site at the same time or the VS1 phone system will stop working.



T1 Status screen

T1 Status Screen

The **T1 Status Screen** is divided into four different tables. The first table (at the top of the status screen) contains columns that correspond to each unassigned T1 channel. The top row of the table lists the last two digits of each port number from lowest to highest. (Reserved ports do not appear in this table.)

Port	98	99	00	02	03	04	06	07	08	10	11	12	14	15	16	18	19	20	22	23	24	26	27	28
DTMF	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
MF	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
CP	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
RSig	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab
XSig	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab	ab

DTMF, MF, CP, Rsig, Xsig status

The second row—DTMF (Dual Tone Multi-Frequency)—lists the last state of DTMF for each channel. The DTMF state is listed in a two-digit code. The following list contains DTMF codes and their definitions.

DTMF Code	Definition
00	DTMF tone D active
01	DTMF tone 1 active
02	DTMF tone 2 active
03	DTMF tone 3 active
04	DTMF tone 4 active
05	DTMF tone 5 active
06	DTMF tone 6 active
07	DTMF tone 7 active
08	DTMF tone 8 active
09	DTMF tone 9 active
0A	DTMF tone 0 active
0B	DTMF tone * active
0C	DTMF tone # active
0D	DTMF tone A active
0E	DTMF tone B active
0F	DTMF tone C active
1x	DTMF tone x inactive

The third row—MF (Multi-Frequency)—lists the state of the last tone heard on each channel. The MF state is also listed in a two-digit code. The following list contains MF codes and their definitions.

MF Code	Definition
00	MF tone 0 active
01	MF tone 1 active
02	MF tone 2 active
03	MF tone 3 active
04	MF tone 4 active
05	MF tone 5 active

06	MF tone 6 active
07	MF tone 7 active
08	MF tone 8 active
09	MF tone 9 active
0A	MF tone KP (Key Pulse) active
0B	MF tone ST (Start) active
0C	MF tone STP active
0D	MF tone ST2P active
0E	MF tone ST3P active
10	MF tones inactive

The fourth row—CP (Call Progress)—switches between one of six states for each channel. The CP state is listed in a two-digit code. The following list contains CP codes and their definitions.

CP Code	Definition
01	Silence
02	Busy
03	Ring
04	Dial Tone
05	Unknown (Voice)
06	Loud

The fifth row—RSig (Received Signal) and the last row—XSig (Transmitted Signal)—are based on the signaling used on that channel.

A = Signal bit A is inactive
A = Signal bit A is active
B = Signal bit B is inactive
B = Signal bit B is active
X = Signal bit B can be active or inactive, the normal signal value is shown in parenthesis

Rsig (Received Signal)

State	E&M	Loop Start	Ground Start
Idle	ax(ab)	aB	Ax(AB)
OffHook	AB		aB
Wink	AB		
Ringing		ab	ab

Xsig (Transmitted Signal)

State	E&M	Loop Start	Ground Start
Idle	ab	aB	aB
OffHook	AB	AB	AB
Wink	AB		
Ringing			

The second table on the T1 status screen contains a series of alarms on the T1 Interface Card. It contains three rows:

Receive—displays status of errors as a signal is being received from the Service Provider.

Transmit—shows the status of errors as a signal is being transmitted to the Service Provider.

Line—shows the general status of the T1 span.

	RAI	AIS	LOS	LOF	PD	B8ZS
RECEIVE	0	0	1	1	1	0
TRANSMIT	1	0		0		
LINE	0	0	1	1		

Alarm status table

The Alarm status table contains the following columns:

RAI—(Remote Alarm Indicator) (yellow alarm) is transmitted by the Service Provider to report a remote trouble condition.

AIS—(Alarm Indication Signal) (blue alarm) is transmitted to the Service Provider to report a local trouble condition.

LOS—(Loss of Signal) occurs when the T1 Interface Card is not receiving a clock signal from the Service Provider.

LOF—(Loss of Framing) occurs when the T1 Interface Card is not receiving framing from the Service Provider or the framing mode on the T1 Interface Card is set incorrectly.

PD—(Pulse Density violation) indicates that the signal contains excessive zeros.

B8ZS—(Binary 8 Zero Substitution) a signal switches between 0 and 1 any time B8ZS line coding is selected to indicate that this setting is active.

Each column contains either a 0 or 1:

0 = Normal operation
1 = Alarm condition

The third table (located to the right of the Alarm Status table) displays the following information:

DSP 1 WD and **DSP 2 WD**—The watchdog timers should always be set to zero.

Valid Intervals—Lists the number of valid 15-minute intervals in 24 hour timeframe.

Current Interval—Lists the number of seconds expired in the current 15 minute interval. It resets every 15 minutes.

DSP 1 WD	0
DSP 2 WD	0
Valid Intervals	96
Current Interval	517

Intervals table

The last table is split into two parts and is located below the Alarm status table. It is divided into two columns:

Current—Lists the current 15-minute interval set of counters. It resets every 15 minutes.

24 Hours—Lists the count for the previous 24 hours.

Each row in the table indicates one of 10 different errors. The first four rows indicate that frames of data with errors in them were received:

ES—Indicates errored seconds.

BES—Indicates bursty errored seconds.

SES—Indicates severely errored seconds.

SEFS—Indicates severely errored framing seconds.

The remaining rows in the table are as follows:

UAS—Indicates that the entire span is an unavailable error.

CSS—Indicates that the clock being received from the Service Provider is not stable. Changes backward or forward in time cause this error.

DMS—Indicates degraded minutes.

PCV—Indicates that a path code violation has occurred.

LES—Indicates line error seconds.

LCV—Indicates line code violations.

	Current	24 Hours		Current	24 Hours
ES	0	0	CSS	517	86400
BES	0	0	DMS	0	0
SES	0	0	PCV	0	0
SEFS	517	86400	LES	517	86400
UAS	517	86400	LCV	0	0

Errors table

T1 Interface Card Checklist

- ❑ Order T1 service from the local phone company based on the needs of the site.
- ❑ Set the switches on the T1 Interface Card to their correct positions.
- ❑ Install the T1 Interface Card in an available ISA slot in the TVS.
- ❑ Connect the PCM bus ribbon cable to the T1 Interface Card and then replace the TVS cover.
- ❑ Connect a RJ-48 cable from the Telco Demarc to the T1 Interface Card.
- ❑ Program the T1 Interface Card using the T1 Edit program in Tel-Site.
- ❑ In the VS1 Editor configuration program, select the **Adapter Setup** pane and select the ports on the PEU that correspond to the switches set on the T1 Interface Card.
- ❑ In the **Port Configurations** pane, assign each T1 Port as a CO Line.
- ❑ Configure each T1 Port for DID or DNIS.
- ❑ In the **DID Routing** pane, set up a DID or DNIS Routing Table.
- ❑ Issue the **`t1 stat [n]`** command at the TVS Command prompt in the **Terminal** window of Tel-Site to verify diagnostic information.

GLOSSARY OF TERMS

This glossary contains an explanation of common telephony and computer telephony features, based on industry-standard functionality and how they relate to the Telecor VS1 telephone system. *Newton's Telecom Dictionary–14th Edition* was used as a basis for some of the definitions.

A

ACD—An abbreviation for Automatic Call Distributor. See *Automatic Call Distributor*.

ACD Status Board—Provides a display of selected ACD groups and their status. It enables supervisors to monitor the progress of calls through an ACD and make necessary changes quickly to ACDs requiring additional agents. The ACD Status Board can be customized to display a single ACD group, scroll through all active ACD groups, or display no ACD groups.

Account Code—A code assigned to a customer or project. Many service companies, such as law offices, engineering firms, and advertising agencies use account codes to track costs and bill clients accordingly. On the Telecor VS1 phone system, Account Codes enable you to place account information in the Station Message Detail Recording (SMDR) records of external calls. Then a call accounting program is used to organize the SMDR information into usable data.

Activation Keys—Used to activate certain features on the Telecor VS1 phone system. They are linked to the Telecor Voice Server (TVS) System ID Number (OHA Serial Number) and only activate software installed on that TVS. The software options that require Activation Keys include: Voice Channel Activation, Automatic Call Distribution (ACD) Agent Package, and Telecor Attendant and Telecor Connect CTI client applications.

ANI—An abbreviation for Automatic Number Identification. See *Automatic Number Identification*.

Announced Transfer—A telephone system feature that provides the ability to transfer a call from one extension to another with an announcement of the call.

Attendant Messaging—The ability for the receptionist to type a message and transfer it with each call. That message appears on the Display Phone or CTI application main screen of the transfer destination.

Auto Hold—Automatically places an active call on hold when a user connects to another call.

Auto Paging—A feature that enables the phone system to automatically announce calls ringing at a specific extension.

Auto Wrap-Up—A feature that allows wrap-up time at the end of a call to complete necessary paperwork concerning the call. The time allowed (in seconds) is set up in the **ACD[n] Config** window. If there is no need for the wrap-up mode function after a call, it can be cancelled by pressing a customized station Feature button or toolbar button.

Automated Attendant—Also called Auto Attendant. A feature that answers calls and plays digital recordings to callers that enables callers to route themselves to an extension through touchtone input, in response to the digital voice prompts. The Telecor VS1 phone system has 20 built-in Auto Attendants—three of which are configured and working Auto Attendants—and 10 sample Auto Attendants. The 17 remaining built-in Auto Attendants can be configured to meet customer requirements.

Automatic Number Identification (ANI)—Reports the number of the calling party. ANI is similar to Caller ID on an analog line. However, the only information provided is the number from which the caller is calling. The Service Provider does not send name information. You can select this option when ordering T1 service.

Automatic Call Distributor (ACD)—A feature used by companies with a high number of incoming calls (such as order taking, dispatching, help desks). An ACD performs four functions: 1) It recognizes and answers an incoming call. 2) It checks its database for instructions on what to do

with the call. 3) Based on these instructions, the ACD sends the call to a recording, such as "somebody will be with you soon, please don't hang up" or to a Voice Response Unit (VRU). 4) It sends the call to an agent as soon an agent is available. There are 10 ACDs available on the Telecor VS1 phone system. A maximum of 96 agents can be logged on to the 10 ACDs, and a maximum of 30 agents can be logged on to an ACD queue at any given time.

Away from Desk—This feature alerts you to incoming calls if you are away from your desk but within hearing distance.

B

Background Music—This feature enables music to be played through speakers in the ceiling and/or speakers in each phone on the system throughout the office.

BLF—An abbreviation for Busy Lamp Field. See *Busy Lamp Field*.

Blind Call Processing—Features that enable you to process a call without answering it.

Blind Disconnect—A feature that enables you to disconnect a call without answering it.

Blind Hold—A feature that enables a user to place a call that is ringing at their station on hold without answering the call.

Blind Transfer—A feature that enables a user to send a call that is ringing at their station to another extension or Voice Mail without answering the call.

Bring to Top on Ring—A feature that brings the Attendant or Connect main screen to the top of a user's screen when a call is received while the user is working in another application.

Busy Lamp Field (BLF)—The Busy Lamp Field is a visual status display for all or some extensions. The BLF indicates if a phone is busy, ringing, or on hold. The Busy Lamp Field console is typically attached to or a part of the user's phone. Traditionally, the BLF console is a device with rows of LEDs showing the status of the extension with solid or blinking lights.

C

Call Bar—A scaled-down version of the Attendant or Connect main screen used for call processing.

Call Forwarding—A feature that enables a user to send incoming calls directly to another station or to Voice Mail.

Call Merging—A feature of Attendant that merges an external call with a second external call.

Call Pickup—A feature that enables users to answer calls ringing, or holding at other extensions.

Call Reporting—A feature that provides a summary of calls and call activity. This summary can include details such as extension number, line number, Caller ID, time of call, duration of call, dropped or lost calls, and so on.

Call Routing—The way in which a call is routed through a particular network, such as a PBX system network.

Call Screening—1) A PBX feature that looks at the digits dialed by the caller to figure out whether the call should be completed. 2) A receptionist or secretary that answers a call and determines who the caller is and announces them to the called party prior to transferring them. 3) A feature on the Telecor VS1 phone system that enables messages sent by the PBX or Attendant to display on station options that have display capabilities. This feature is useful for station users to determine if a call is important enough to answer.

Call Waiting—Call Waiting is a feature of phone systems that indicates someone is trying to call you. You might hear a beep or see a light on the phone or a message on the CTI display.

Caller Database—A database in Attendant that contains names and numbers of all outside callers. As names are added to the Caller Database, Attendant fills the **Calls** list of the **Transfer** window with the names of people who have called from that number in the past. Attendant then fills in the **Transfer To** list with the names of people with whom that caller has spoken in the past.

Caller ID—A service that the local phone company provides, usually called CLASS. The information about who's calling and/or their phone number is passed to your phone between the first and second ring of the incoming call.

Caller ID Option Module—Hardware that must be installed in the PEU for Caller ID to work on the Telecor VS1 phone system.

Caller ID Routing—Also called Call Routing based on Caller ID or Intelligent Call Routing. Caller ID Routing enables you to target incoming calls to specific extensions based on Caller ID. You can set up Caller ID Routing for up to 800 entries on the Telecor VS1 phone system.

Calls List—A feature of Attendant and Connect. The **Calls** list displays all active or present internal and external calls at a user's station and their status. The Call Info, Number, Time and Hold columns provide visual information about each call including: Text Messages sent by the receptionist or Auto Attendant, Name and Number Caller ID (if available), length of each call, and length of each call on hold.

Central Office—A telephone company facility where subscriber lines are joined to switching equipment to connect with other subscribers.

Centrex—A business telephone service offered by a local telephone company from a local central office. Centrex includes features such as intercom, call forwarding, and call transfer.

CO—An abbreviation for Central Office. See *Central Office*.

CO Flash—A signal provided by the Telecor VS1 phone system to the Telephone Company or Central Office indicating that special instructions will follow.

CO Lines—These are the lines connecting the phone system to the local telephone company's central office which in turn connects to the nationwide telephone system (network).

Command Scripts—Automated processes for executing one or more commands. The TVS runs Command Scripts much like DOS runs batch files. Telecor VS1 phone system Command Scripts provide a convenient method for controlling the various aspects of system operation. A Command Script consists of one or more PBX commands that control the system TVS.

Computer Telephony—Computer telephony is the adding of computer intelligence to the making, receiving, and managing of telephone calls. Computer telephony encompasses six broad elements: Messaging, real-time connectivity, transaction processing and information access through the phone, adding intelligence to phone calls, technologies, and new standards.

Computer Telephony Interface Module (CTIM)—The Telecor Computer Telephony Interface Module (CTIM) provides a digital interface for a computer using Attendant Connect to the Telecor VS1 phone system. The CTIM is an external device connected by a 9-pin serial cable to an available COM port on the computer.

Conference Rooms—Several parties can be added to a phone conversation through conferencing. The Telecor VS1 phone system has three 16-party Conference Rooms. They can operate simultaneously, and can include any combination of inside or outside parties for a total of 16 participants.

Consultation Hold—A PBX feature that enables an extension to place a call on hold while speaking with another call.

CTI—An abbreviation for Computer Telephony Integration. See *Computer Telephony*.

CTIM—An abbreviation for Computer Telephony Interface Module. See *Computer Telephony Interface Module (CTIM).*

Cut-Over Box—Provides power failure transfer for four CO lines. If a power failure occurs, each station connected to the Cut-Over Box becomes a direct outside line for incoming and outgoing calls.

D

DDE—*See Dynamic Data Exchange*

DCU—An abbreviation for Dry Contact Unit. See Telecor *Dry Contact Unit*.

Dial By Name—A feature that enables you to dial someone by spelling their name on a touchtone keypad. The caller inputs the appropriate digits or letters. When the system recognizes a match, a recorded announcement states the name of the dialed party for confirmation by the caller before automatically completing the call. If the input digits are not uniquely associated with a particular station, the system may ask the caller to pick a name from a menu of choices. The Telecor VS1 phone system can be programmed for Dial By Name using either the first three letters of a person's first name or the first three letters of a person's last name.

Dialed Number Identification Service (DNIS)—A feature of 800 number lines that provides the number the caller dialed to reach the attached system. DNIS tells you the number that the caller dialed. Using DNIS capabilities, one trunk group can be used to serve multiple applications. The DNIS number can be provided in a number of ways, in-band or out-of-band, ISDN or through a separate data channel. Generally, a DNIS number is used by the answering system to identify the reason the caller dialed.

DID—An abbreviation for Direct Inward Dialing. See *Direct Inward Dialing*.

Digital Promotion Recording—Informs customers about new products, sales and promotions. The recording can be changed as needed. No external hardware is required. All Telecor VS1 phone system station options enable you to record promotional recordings.

Direct Inward Dialing (DID)—The ability for a caller outside a company to call an internal extension without having to pass through an operator or Automated Attendant. DID Routing on the Telecor VS1 phone system can only be used with a Telecor T1 Interface Card, and T1 service must be ordered from your local Service Provider.

Direct Inward System Access (DISA)—Enables a user to gain access to the Telecor VS1 phone system internal dial tone from an external phone not connected to the system. DISA allows feature options such as running Command Scripts, dialing extensions on the system, and access to a CO port to make external calls.

Direct Station Select (DSS)—A phone system feature that enables a user to press a programmed button on their phone to dial another extension. Typically, DSS is a part of Busy Lamp Field (BLF), which shows the status of that extension as well (busy, ringing, holding).

DISA—An abbreviation for Direct Inward System Access. See *Direct Inward System Access*.

Disconnect—The breaking or release of a circuit connecting two telephones or data devices. A Disconnect signal is sent from one device to the other to shut down the connection.

Distinctive Ring—Users can choose from 12 different rings and eight different volume settings on the Telecor Display Phone Model 200 (DP200). Telecor CTI applications also enable users to choose a distinctive ring for incoming call notification.

DND—An abbreviation for Do Not Disturb. See *Do Not Disturb*.

DNIS—An abbreviation for Dialed Number Identification Service. See *Dialed Number Identification Service*.

Do Not Disturb (DND)—Calls do not ring through to stations placed on DND. Calls to Telecor VS1 stations on DND are automatically transferred to Voice Mail.

DP200 Display Phone—A phone designed for use with the Telecor VS1 phone system. It is equipped with a 2 line by 16 character display, which provides information about the calls being handled. The DP200 Display Phone has the capability to process five calls.

Dry Contact Unit (DCU)—The Dry Contact Unit is a hardware interface to which external contacts are connected in systems that use Model 250 Port Expansion Units. The Dry Contact Unit provides connections for two speakers, two music inputs, and two dry contact outputs.

DSS—An abbreviation for Direct Station Select. See *Direct Station Select*.

DTMF—An abbreviation for Dual Tone Multi-Frequency. See *Dual Tone Multi Frequency*.

Dual Tone Multi-Frequency (DTMF)—A term describing push button or touchtone dialing. When you touch a button on a push button keypad, it makes a tone and sends an in-band signal.

Dynamic Data Exchange (DDE)—A Microsoft®-defined method for exchanging data between Windows® applications. Support for DDE must be explicitly coded into an application in order for this feature to be supported; it is not automatically included in Windows service. Data is passed between DDE clients (or "sources") and DDE servers (or "destinations") as "messages". Connect can be both a DDE client and a DDE server.

E

Emergency Boot Floppy—A floppy disk that can be used to restart the TVS in case of a system failure. Before making changes to the TVS, it is recommended that you make an Emergency Boot Floppy.

Executive Privilege—Enables an extension to monitor conversations on other extensions.

Extension—A station option (telephone or CTI application) connected to a port on the PEU. Also refers to the digits dialed to gain access to a station option, voice mail, guest mailbox, work group, Auto Attendant, Paging Zone, conference room, or ACD.

Extensions List—A feature in Attendant and Connect that contains a list of all names and extensions on the Telecor VS1 phone system.

External Call—A call made from or made to a number outside the Telecor VS1 phone system.

External Contacts—The PEU Model 205 and DCU Model 100 each have six external contacts that are used to connect Zone Pager Outputs such as overhead paging systems; Music Inputs such as audio sources; and Relay Outputs such as electric door locks.

External Flash—A signal that is sent outside the phone system.

F

Flash—A signal sent to a PBX or Centrex indicates DTMF or in-band instructions will follow, such as transferring a call to another extension.

Flexible Ringing Assignments—The capability to program a CO line to target multiple destinations, such as a receptionist, a group of extensions, Voice Mail, and so on.

Flexible Station Numbering—The capability to program station extension numbers with the number of digits desired. For example three-digit extension numbers or four-digit extension numbers. Each station on a system must have the same number of digits.

Forward—A feature that temporarily redirects incoming calls to another extension.

Forward to an Outside Number Through an Automated Attendant—Redirects calls to a selected outside number. This feature is used primarily during non-business hours.

Function Codes—Two-digit codes that are entered when programming features using the Function Key Programmer.

Function Keys—Twelve default Function Keys enable DP200 Display Phone users to gain access to Telecor VS1 phone system features such as Call Forwarding, Paging, Transfer, and so on. Default settings can be changed through the configuration program in the **Function Key (System)** window, or by dialing **7801** from a Telecor Display Phone Model 200 (DP200).

G

Group Ring—A feature that enables incoming calls to simultaneously ring a group of extensions.

Guest Mailboxes—Virtual ports for individuals without a station option connected to the Telecor VS1 phone system. Guest Mailboxes are assigned an extension and can gain access to all Voice Mail features. You can gain access to Guest Mailboxes from any internal station option, or from an outside line. Up to 200 Guest Mailboxes can be configured.

H

Handsfree Intercom—A feature that enables users to program their phone to automatically answer all internal calls through the speakerphone, without touching a button or picking up their handset. Handsfree must be selected from the **Port [n]** window of the TVS for this feature to work.

Hold—Suspends a call without disconnecting.

Host Adapter—Installed inside the TVS and used to connect PEUs to the TVS.

Hybrid System—A term used to describe a phone system that has attributes of both key systems and PBXs. One distinguishing feature is that a hybrid system can use normal single line phones in addition to the normal proprietary telephones. Another distinguishing feature is that it's non-squared. In other words, not every trunk appears as a button on every phone in the system, as is the case with key systems.

I

Incoming Call Notification—A feature of Attendant and Connect that enables a user to determine how to be notified when a call is received.

Initial Dialup Delay—A user-defined amount of time Tel-Site needs before making the connection to a remote site TVS modem. The delay required may need to be adjusted according to location.

Internal Call—A call made from or made to another extension on the Telecor VS1 phone system.

Internal Party ID Numbers—The capability to display the extension number of the internal calling party on phone displays and CTI screens.

ISDN—An abbreviation for Integrated Services Digital Network. ISDN is an evolving communications standard available at 144,000 bps for desktop applications (BRI) and 1,544,000 bps for telephone switches, computer telephony and voice processing systems (PRI).

K

Key System—A system in which the telephones have multiple buttons permitting the user to directly select a telephone line.

Keypad—A feature of Attendant and Connect enabling you to use the mouse to click the numbers on the keypad. When the Keypad is active, you can use the computer keyboard to type 0–9, *, # and F for flash.

L

LCR—An abbreviation for Least Cost Routing. See *Least Cost Routing*.

Least Cost Routing (LCR)—A phone system feature that automatically chooses the "least cost" long distance method to dial a long distance call. Least Cost Routing is typically set up as a table in phone system configurations.

Line Pool—A specific group of lines designated to make external calls.

Line Queuing—A feature that places a call in a queue to wait for the next available line. This is different than ACD Queue, which waits for the next available agent.

Loop Tapes—One to 12 Loop Tape messages can be created for use on the Telecor VS1 phone system. Their use is limited by the number of voice channels available on the system (one Loop Tape per voice channel). Each Loop Tape created permanently uses the resources of one voice channel. Whether the Loop Tape is in use or not, it plays continuously during system operation.

M

Music-on-hold—Also referred to as Promotions-on-hold; music or recorded promotions heard by callers when they are placed on hold. Most phone systems require the connection of peripheral equipment to accommodate on-hold music or recordings. The Telecor VS1 phone system comes configured with a generic music-on-hold file.

Mute—A feature that disconnects the handset microphone or speakerphone microphone during a connected call.

N

NANP—An abbreviation for North American Numbering Plan. See *North American Numbering Plan*.

Night Mode—The way that calls are processed after normal business hours. Typically, night mode allows for a different greeting or for calls to ring to an Auto Attendant.

North American Numbering Plan (NANP)—The method of identifying telephone lines and area codes in the public network of North America.

O

Off-Premises Extension (OPX)—A phone that is a part of the phone system located in a different building or location away from the main premises (such as a security booth at an entrance to the parking lot, or a warehouse located next to the main building). The OPX is connected to the system with a special OPX phone line. The extension acts just like an extension on the system.

OPX—An abbreviation for Off-Premises Extension. See *Off-Premises Extension*.

P

Paging (Overhead Paging)—The Telecor VS1 phone system enables users to dial a paging zone from any station option, and make an announcement through the handset/headset microphone.

Paging (Pager Notification or Voice Mail Notification)—A Voice Mail feature where one to three digital pagers can be paged upon receipt of a new Voice Mail message.

Paging Zones—The Telecor VS1 phone system has 20 software Paging Zones and two hardware Paging Zones available. The software Paging Zones enable you to page over the DP200 Display Phone speakerphone, and the two external Paging Zones.

Park—The Telecor VS1 phone system has 10 Park Zones in which users can park calls. Parked calls can be retrieved from any station on the phone system.

PBX—An abbreviation for Private Branch Exchange. A phone system that is considered a small version of the phone company's larger switching office. A PBX requires dialing a "9" to connect to an outside CO line. PBXs are typically found in medium to large-sized businesses with wide spread use of single line phones. PBXs provide the advantage of telephone lines pooled and shared by all employees.

PC Option Module (PCOM)—The Telecor VS1 phone system PC Option Module (PCOM) provides an externally connected serial interface to a computer for data sent by the Telecor Voice Server (TVS), and enables a regular phone set to remain on the desk.

PEU—An abbreviation for Port Expansion Unit. See *Port Expansion Unit*.

Port—The physical interface on a phone system where connection of CO lines, telephone stations, faxes, modems, and so on, are connected.

Port Expansion Unit (PEU)—The hardware interface to which CO lines and stations are connected. The Model 200 and 205 PEUs have 16 ports for connecting CO lines or stations. The first eight ports can be configured as either CO ports or stations. The remaining eight ports can only be configured as stations. All 16 of the Model 250 PEU ports can be configured as CO lines or stations, when used with Version 2.9 or higher of the VS1 software.

Power Failure Transfer—Enables specific telephones on a phone system to receive incoming calls in the event of a power failure. On most systems, designated telephones must be single line telephones.

Primary Loop Tape—The Loop Tape callers hear first when calling an Auto Attendant. Each caller hears the Primary Loop Tape from the beginning. If the Primary Loop Tape is being played when a new call arrives, the new caller hears ringing until the loop starts over.

Private Branch Exchange—see *PBX*.

Private Line Assignment—The capability to dedicate a CO line to ring a specific extension. For example, an executive may want to have their own dedicated phone line that rings directly to their telephone.

Programmable Relays—Relay contacts that enable the connection of devices such as security locks which can be activated through the system or at a station.

R

Remote System Access (RSA)—The capability to gain access to phone system programming functions from a remote location through DISA or through a modem.

Ring Over Pager—The capability for a phone system to ring incoming calls over paging speakers. Typically this is used after normal business hours or in heavy industrial, high-noise applications.

RSA—An abbreviation for Remote System Access. See *Remote System Access*.

S

Scheduled Events—An event command set up in system programming that includes a specific day of the week, time of the day, and the action the phone system is to perform. Typically used for switching to Night Mode at a certain time of the evening.

Secondary Loop Tape—The Loop Tape callers hear repeatedly after the Primary Loop Tape until they leave the ACD Queue.

SMDR—An abbreviation for Station Message Detail Recording. See *Station Message Detail Recording*.

Speed Dials—A telephone number programmed into the system or a specific telephone that is automatically dialed when a feature button is pushed or special code dialed. System Speed Dials are accessible system-wide and have a two-digit number (01-99) that users press to dial the specific number. Personal Speed Dials are accessible only from the extension where they were created and have a one-digit number (0-9) that users press to dial the specific number.

Station Message Detail Recording (SMDR)—A feature that records call information about telephone call activity such as time, date, length of call, Account Codes, and so on. SMDR is usually used with a call accounting program that organizes the SMDR raw data into usable information.

Station Status Window—A feature of Connect that enables you to monitor and instantly dial other stations on the Telecor VS1 phone system.

System Key—Required to upgrade the TVS software. The System Key is specific to the system being upgraded and must be requested through Telecor Technical Support. No valid key means no voice channels and therefore no Auto Attendants or Voice Mail.

T

T1—A digital transmission link with the capacity of 1.544 Megabits per second (Mbps). T1 uses two pairs of normal twisted wires, and can normally handle 24 simultaneous voice conversations.

T1 Interface Card—Installed in the Telecor Voice Server (TVS) and supports 24 T1 channels. Two T1 Interface Cards can be installed in the Telecor VS1 phone system.

TAPI—An abbreviation for Telephone Applications Programming Interface. A standard developed by Microsoft® and Intel®, which provides a programming interface for Windows® operating systems. It enables users and application developers to take advantage of telephone services and capabilities through the computer.

Telco—An abbreviation for Telephone Company in reference to a local Telephone Company.

Telecor **Voice Server (TVS)**—The central component of the Telecor VS1 phone system. It houses the proprietary software and hardware that operates the system.

Telephone Applications Programming Interface—See *TAPI*.

Telephony Services Application Programming Interface—See *TSAPI*.

Tel-Site System Management Application—An application that enables Telecor VARs to make remote Configuration changes to customer sites.

Toll Restriction—A feature that restricts specific phones from making certain types of calls, such as long distance calls.

Transfer—A feature that enables a user to send a call to another extension or to Voice Mail. The call does not need to be answered to transfer.

Trunk—A communication line between two switching systems. A switching system includes PBX's and equipment in Central Offices (CO).

TSAPI—An abbreviation for the Telephony Services Application Programming Interface standard developed by Novell® and AT&T®. TSAPI provides a programming interface for Novell networks and is designed to take advantage of a network server to provide telephony functions. TSAPI requires a dedicated voice server or serial LAN connection from the station to the PBX.

TVS—An abbreviation for Telecor Voice Server. See Telecor *Voice Server*.

V

Voice Channel—A voice channel is a Telecor feature that enables you to play and record voice files. A maximum of 12 voice channels are available.

Voice Mail—Voice Mail enables you to leave, send, and forward messages to individuals who are not answering their phone. Voice Mail is assigned to each station port. The Telecor VS1 phone system integrates Voice Mail into the system.

Voice Mail Escalation—Feature that enables Voice Mail messages to be forwarded to up to 20 other Voice Mail extensions.

Voice Mail Notification—see *Paging (Pager Notification or Voice Mail Notification)*.

W

Work Groups—Virtual extensions assigned to groups of extensions on the Telecor VS1 phone system. Work group stations ring simultaneously when a call is sent to the work group extension. Up to 20 Work Groups containing 20 extensions each can be programmed on the Telecor VS1 phone system.

Z

Zone Paging—The ability to page a specific area in a building. Typically, departments are put into zones in system programming to facilitate this type of paging.

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